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DATE: Tuesday, February 24, 2004

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☐	L16	L12 and l6	0
☐	L15	L12 and l5	1
☐	L14	l12 NOT l13	44
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☐	L12	19991215	54
☐	L11	packet near8 (associate or associated or associating) near8 transmission near8 priority	114
☐	L10	packet near8 transmission near8 priority	1700
☐	L9	L8 and l2	3
☐	L8	bitstream same network	923
☐	L7	bitstream same network same (differential near8 service)	0
☐	L6	object-based adj3 media	5
☐	L5	19991215	4
☐	L4	L3 and l2	11
☐	L3	transmission near8 priority near8 level	977
☐	L2	differential near8 service	1740
☐	L1	differential near8 service near8 transmission near8 priority	0

END OF SEARCH HISTORY

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L6: Entry 3 of 5 File: PGPB Feb 28, 2002

DOCUMENT-IDENTIFIER: US 20020024539 A1
TITLE: System and method for content-specific graphical user interfaces

Detail Description Paragraph:
[0024] An exemplary embodiment of the present invention is described herein using an MPEG-4 standard system. MPEG-4 is an international standard for the object-based representation of multi-media content, and allows creation of multi-media content with multiple audio, video, image, and text elements. The MPEG-4 Systems standard specifies the technology for on-screen layout, packaging, and playing back mixed media components, and includes an extensible framework for customizing MPEG-4 applications. The capability of MPEG-4 to treat all elements of a multi-media program as individual objects allows for innovative ways of using downloadable and content-specific GUIs.

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L11: Entry 1 of 41 File: PGPB Mar 14, 2002

DOCUMENT-IDENTIFIER: US 20020032001 A1
TITLE: SYSTEM AND METHOD FOR TRANSMITTING DATA

Application Filing Date:
19991001

Summary of Invention Paragraph:
[0002] Consumers have an insatiable appetite for information and entertainment, colloquially referred to as "content." This can be seen in the popularity of television, the internet and other content based media that are delivered to subscribers or users over various "pipelines." A pipeline is a system that transmits data from a content provider, e.g., television station, website on the internet, etc., to a subscriber. For example, internet service providers such as AmericaOn Line use the telephony system as a pipeline to transmit information to its subscribers. The subscribers use a computer modem to dial-in to an internet service provider. Once on-line, the subscribers have access to various content providers (websites) and can download or upload information. Unfortunately, this is often a slow and cumbersome technique for conveying large quantities of data because the telephony system has transmission speed and bandwidth limitations. Certain subscribers have installed specialized high-speed telephonic connections, but the practice is not widespread because of the prohibitive costs.

Summary of Invention Paragraph:
[0003] Similarly, various conventional pipelines deliver video information from content providers with varying degrees of success. Conventionally, television stations use a wireless pipeline for delivering content to users. The television stations simply broadcast signals in a dedicated portion of the electromagnetic spectrum. Users access the signals with roof-top antennas. Cable systems are also used in many areas. These systems use coaxial cable to deliver video with increased quality and quantity directly to a user's home or premises. However, conventional cable systems do not allow for interactive feedback to the content providers over the cable system. Retrofitting the existing cable systems with this feature will be expensive and time consuming.

Detail Description Paragraph:
[0021] Transmission system 10 can transmit audio, video, or other data for use by a computer, a television, a telephone or other appropriate terminal of subscribers 14. Transmission system 10 provides a pipeline between communication service providers 18 and subscribers 14. Communication service providers 18 may, for example, provide services such as video, interactive video, internet connection, telephony or access to other content based services. Transmission system 10 includes head end 17 coupled between communication service providers 18 and central hub 12. Head end 17 can communicate with communication service providers 18 and central hub 12 over any appropriate communication link such as wireless, including satellite and microwave or wired communication link as shown in FIG. 1. Transmission system 10 further includes a number of digital repeaters, represented here by digital repeaters 16a through 16c. It is understood that transmission system 10 includes an appropriate number of digital repeaters referred to collectively as "digital repeaters 16."

Detail Description Paragraph:
[0024] Transmission system 10 uses packets of digital data to increase the number of effective channels of the system. Specifically, transmission system 10 uses MPEG compression so as to transmit as many as 6 video channels in one 6 MHz channel. Similarly, transmission system 10 transmits multiple data channels in a single 6 MHz channel.

Detail Description Paragraph:
[0035] In one embodiment, transmission system 10 of FIG. 1 is controlled by a media access control (MAC) protocol. The MAC protocol supports video-on-demand services, data services, and control functions. The MAC protocol is an asymmetric protocol which uses fixed size MPEG 13818-1 transport stream transport packets for downstream transmissions and short, fixed size slots for upstream transmission. The protocol is consistent with DAVIC Technical Specifications 1.2 for MMDS by using transport packets for downstream. DAVIC presently leaves the upstream undefined ("reserved"). The MAC protocol of this embodiment is described in conjunction with FIGS. 5A through 10.

Detail Description Paragraph:
[0038] Downstream packet delivery is composed of two types of MPEG-2 transport packets, data transport packets and acknowledgment transport packets. Data transport packets are identified by a unique packet identification (PID) and acknowledgment transport packets are identified by a different unique PID. As illustrated in FIGS. 5A and 5B, the downstream data stream is composed of a sequence of 50 data transport packets 200 followed by an acknowledgment transport packet 202.

Detail Description Paragraph:
[0041] FIG. 6 is a diagram that illustrates the frame format of the downstream data packets 200. The basic form of the packet is an MPEG 13818-1 transport packet of fixed size (e.g., 188 bytes) and contains a 4 byte header. In general, transport packet PIDs are used to indicate program content including video and audio streams. In order to prevent any conflict with video programming the PID field is used only in the most limited manner. Specifically, two PIDs are used, one to indicate "normal" downstream data and the other for downstream acknowledgment.

Detail Description Paragraph:
[0052] 6) The priority field is 1 bit that indicates the message priority to headend 17.

Detail Description Paragraph:
[0085] 2. A single bit in the upstream packet could be used to distinguish user data from control data. A bit that is presently unused and could be used for this function is the priority bit.

Detail Description Paragraph:
[0094] Other timers may be required to implement non-essential components of the protocol, e.g., support for the priority scheme. These are not described here.

Detail Description Paragraph:
[0095] The protocol also provides a limited facility for expedited data upstream. The priority field in the upstream packet is used by the subscriber to indicate to headend 17 that the data in the packet should be expedited. This information may be used or ignored by headend 17 depending on the upstream algorithm implemented in headend 17.

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L11: Entry 25 of 41 File: USPT Aug 15, 2000

DOCUMENT-IDENTIFIER: US 6104757 A
TITLE: System and method of error control for interactive low-bit rate video transmission

Abstract Text (1):
A new retransmission-based error control technique that does not incur any additional latency in frame playout times and is suitable for interactive video applications. This retransmission technique combined with layered video coding yields good error resilience for interactive video conferencing. The technique exploits the temporal dependency of inter-coded frames and can be easily incorporated into motion-compensation based coding standards such as MPEG and H.261, achieving very good compression efficiency.

Application Filing Date (1):
19980515

Brief Summary Text (5):
Most standard video compression coding schemes, such as H.261, H.263, and MPEG, are not designed for real-time transmission over a lossy packet switched network, but primarily for video storage, e.g., CD or VHS tape. Although these schemes are capable of high compression efficiency, packet loss severely degrades video reception quality. This degradation is due to motion compensation techniques used by coders and decoders (hereinafter "codecs") for removing temporal redundancy in video streams. Motion compensation removes inter-frame temporal redundancy by encoding only a pixel difference or a prediction error between a current image and a previously transmitted image often referred to as a reference frame. A single occurrence of packet loss can introduce an error in a reference frame, which can propagate and get amplified in subsequent frames as more packets are lost.

Brief Summary Text (7):
Retransmission-based error recovery (REC) provides improved error resilience without incurring significant bandwidth overhead because packets are retransmitted only when they are indicated lost. Retransmission does involve transmission delays, however, and has been known to be ineffective for interactive real-time video applications such as internet video conferencing. Many have proposed the use of extended control or playout times to allow retransmitted packets to arrive in time for display. This implies that the playout time of a frame is delayed by at least three one-way trip times after initial transmission, two for packet transmissions and one for a retransmission request. Under current internet conditions, such a delay would be intolerable for interactive video applications.

Brief Summary Text (12):
In the present invention, retransmitted packets do not have to arrive in time for display to be useful. The construction of motion-compensated codecs such as H.261 and MPEG reveals the correct image reconstruction of a current playout depends on successful receipt of previous reference frames. Thus, although a frame may not arrive in time for its display (due to loss or delay), the lost or delayed frame is still useful for reconstruction of successive frames, whose playout time is later than the delayed frame. In other words, if a packet is lost, then the packet can be retransmitted and arrive after the frame that the packet belongs to is played out.

If the retransmitted packet arrives before the reconstruction of subsequent frames, however, the distorted frame can be repaired and used as a clean reference frame for the next frame thereby preventing errors in the reference frame from propagating forward.

Brief Summary Text (16):
Retransmission schemes for distributing MPEG-coded video over a best-effort network such as the internet have been used in the recovery of lost packets in a video multicast transmission. By transmitting different frame types (I, P and B frames) of MPEG to different multicast groups, a simple layering mechanism was implemented in which a receiver can adjust frame playout times during congestion by joining or leaving a multicast group. For instance, consider a MPEG picture pattern: IBBPBBPBBPBB. By delaying the playout time of each frame for one frame interval, the playout time of a frame is extended by one frame interval. This delayed playout time is termed the adaptive playback point. If a receiver leaves the B frame group because of congestion, the adaptive playback point is additionally extended by three frame intervals. In other words, a P frame can be displayed after three frame intervals from its reception. The scheme has been shown effective for non-interactive real-time video applications. This technique, however, may not be useful for interactive real-time video applications because of the possibility of long playout times.

Brief Summary Text (18):
A forward error correction scheme known as priority encoding transmission (PET) has been applied to hierarchically encoded MPEG video. A temporal layering scheme was used in which reference frames (e.g., I frames and P frames) were given a higher priority than other temporally dependent frames (e.g., B frames). Since B frames are temporally dependent on P and I frames which are more reliably received, this technique effectively suppresses error propagation. However, in low bit rate video conferencing, the frequency of I and P frames must be kept very low because of their low compression efficiency (typically 3 to 8 times lower than that of B frames). Thus, the resulting images can be very jerky as packets are being dropped to affect B frames. If the frequency of I and P frames has to be increased, then the amount of redundant bits added by PET also increases.

Brief Summary Text (19):
A priority packetization scheme has been applied to an MPEG encoded video stream for ATM transmission. A frequency truncation technique was applied in which a fixed number of DCT coefficients of each DCT block are allocated to the HP (high priority) data. It was shown that by utilizing this type of packetization, basic image quality can be maintained if the HP stream is guaranteed to be received. However, it does not solve the error propagation problem because essential signals of a frame are still temporally dependent on both the essential and enhancement signals of its reference frame. Since the decoded enhancement signals are more often erroneous, frames that depend on enhancement signals can perpetuate the same error.

Brief Summary Text (21):
In quality assurance layering (QAL) errors due to loss of LP (low priority) packets do not propagate because each frame is temporally dependent only on the essential signals of its reference frame which are assumed to be reliably received. The effect of two different priority packetization techniques similar to frequency truncation and energy threshold has been studied. It was shown that the energy threshold method performs slightly better than frequency truncation when the HP stream uses about 50% of the total bandwidth allocated.

Brief Summary Text (22):
Similar layering techniques to QAL have also been proposed. In such techniques, each frame is first decimated to produce a low-resolution image, then the low-resolution image is coded using H.261 or a DCT-based codec and packetized as HP

data. The original image is then compared with the decoded frame of its low-resolution frame to produce a difference image which is coded using a different codec and packetized as a LP data. This coding scheme will have similar error resilience as the QAL technique since the LP data is temporally dependent only on the HP data. However, this codec could be computationally more demanding because in addition to the initial filtering, two types of encoding are performed on the same frame.

Brief Summary Text (23):
QAL has been applied to video transmission over a mobile network to solve the fading problem commonly experienced during a hand-off period. By keeping the size of the HP stream large (about 83% of the total bandwidth), video quality, even under fading, can be kept comparable to that during normal operation. Priority layering techniques are also applied to still JPEG image transmission. A frequency truncation layering technique that partitions DCT blocks of JPEG encoded frames into essential (consisting of the DC coefficients) and enhancement (consisting of all the AC coefficients) layers has been used. The effectiveness of layered coding through the hierarchical mode of JPEG was studied and yielded a statistical analysis showing the overhead of the coding method can be insignificant.

Brief Summary Text (26):
Many motion compensation prediction-based codecs, such as MPEG, and H.261, are useful for internet interactive video transmission despite previously discussed drawbacks. Some of the drawbacks of motion compensated codecs include computational complexity, error resilience, tight coupling between the prediction state at the encoder and that at the decoder, and compute-scalable decoding. The present invention shows that H.261 equipped with REC schemes achieves comparable error resilience to that of INTRA-H.261, and combined with a priority layering coding technique, yields better video quality than INTRA-H.261. under the same bit rate and loss rate. Other disadvantages can be overcome with relatively simple modifications to the codecs. For instance, compute-scalable decoding is achieved by decoding only periodic frames and shedding off the computational load for decoding non-periodic frames in PTDD. If the distance between two periodic frames is too large, the display image may look too jerky. However, by having several different types of periodic frames, each of which has a different TDD, this problem is overcome.

Detailed Description Text (21):
FIG. 7 illustrates a scheme called periodic temporal dependency distance (PTDD). For PTDD, every i.sup.th frame has an extended TDD of "i" frames (we call this frame a periodic frame) while all the other inter-frames have a TDD of 1. The TDD of periodic frames in FIG. 5 is four. In fact, the pattern of the temporal dependency is very similar to a picture group pattern of MPEG. All frames with a TDD of four can be regarded as P frames while the other frames can be regarded as B frames (except the first frame). Thus, this scheme is easily incorporated into MPEG. PTDD does not incur much computational overhead and does not require many additional frame buffers.

Detailed Description Text (25):
Error resistance for non-periodic frames can be improved by employing a layered coding scheme that packetizes encoded frames into essential (high priority) and enhancement (low priority) signals. Although layered coding was originally developed for a network paradigm such as ATM and RSVP, where a certain amount of bandwidth can be reserved, it can also be used

Detailed Description Text (27):
A version of quality assurance layering (QAL) is shown in FIG. 8. It is a modification of a H.261 encoder and augmented with priority packetization and conditional replenishment. After DCT coefficients are quantized (Q), they are partitioned into HP and LP layers as described in FIG. 9. For a fixed nonzero

integer b less than 64, the first b coefficients are allocated to the HP layer, and remaining coefficients are allocated to the LP layer. The subsequent inter-frame and the currently encoded frame are used to perform motion estimation and conditional replenishment for encoding the next frame. The motion vectors and the HP coefficients of the current frame are used to reconstruct a predicted frame. The difference between the subsequent frame and the predicted frame is encoded. In the scheme, each frame temporally depends only on the essential signals of its reference frame. Since a frame is reconstructed from the essential signals of its reference frame, an error in the enhancement signals of the reference frame caused by packet loss does not carry over. Thus, even if all LP stream packets are discarded, a certain level of video quality can be maintained.

Detailed Description Paragraph Table (1):
LIST OF ACRONYMS USED THROUGHOUT THE TEXT
ATM Asynchronous Transfer Mode CODEC or codec Coder/Decoder DCT Discrete Cosine Transformation FEC Forward Error Correction H.261 A Coder/Decoder Image Compression Scheme H.263 A Coder/Decoder Image Compression Scheme HL.261 A Coder/Decoder Image Compression Scheme HP High Priority HP.261 A Coder/Decoder Image Compression Scheme HPF.261 A Coder/Decoder Image Compression Scheme HPL.261 A Coder/Decoder Image Compression Scheme INTRA-H.261 A Coder/Decoder Image Compression Scheme INTRAL-H.261 A Coder/Decoder Image Compression Scheme JPEG Joint Photographic Experts Group - A Standard Compression Format for Color Images LP Low Priority MPEG Motion Pictures Experts Group - A Compression Format for a Series of Images NACK No Acknowledgment PET Priority Encoding Transmission PSNR Peak Signal-to-Noise Ratio PTDD Periodic Temporal Dependency Distance QAL Quality Assurance Layering REC Retransmission Based Error Control STORM Structure Oriented Resilient Multicast TDD Temporal Dependency Distance

Other Reference Publication (1):
Albanese et al., "Priority Encoding Transmission", IEEE Transactions on Information Theory, vol. 42, No. 6, (written Aug. 1994; published Nov. 1996), pp. 1-34.

CLAIMS:

6. The system of claim 5 further comprising means for packetizing each frame into high priority signals and low priority signals wherein only high priority signal data is used for reconstruction of a frame.

7. The system of claim 3 further comprising means for packetizing each frame into high priority signals and low priority signals wherein only high priority signal data is used for reconstruction of a frame.

8. The system of claim 1 further comprising means for packetizing each frame into high priority signals and low priority signals wherein only high priority signal data is used for reconstruction of a frame.

10. A system for displaying an incoming stream of video images comprising:

(a) means for receiving said stream of video images in the form of frames comprised of packets, each frame based on a reference frame, and each frame having a predesignated playout time;

(b) means for packetizing each frame into high priority signals and low priority signals wherein only high priority signal data is used for reconstruction of a frame;

(c) means for determining whether any frame packets were lost during transmission;

(d) means for requesting that any lost frame packets be retransmitted;

(e) means for determining whether retransmitted lost frame packets have arrived prior to the frame's playout time;

(f) means for reconstructing said frame using the originally transmitted frame packets and, if necessary, said retransmitted lost frame packets, provided said retransmitted lost frame packets arrived before the expiration of the frame's playout time;

(g) display means for displaying the reconstructed frame;

(h) means for storing the just displayed reconstructed frame as said reference frame; and

(i) means for reconstructing said reference frame using the just displayed reconstructed frame and said retransmitted lost frame packets that did not arrive before the expiration of the frame's playout time.

11. A system for displaying an incoming stream of video images comprising:

(a) means for receiving said stream of video images in the form of frames comprised of packets, each frame based on a reference frame, and each frame having a predesignated extendable playout time wherein every i.sup.th, i being a integer, frame has a variable extended temporal dependency distance that can be set to any number of frames while all other frames have a temporal dependency distance of one frame;

(b) means for packetizing each frame into high priority signals and low priority signals wherein only high priority signal data is used for reconstruction of a frame;

(c) means for determining whether any frame packets were lost during transmission;

(d) means for requesting that any lost frame packets be retransmitted;

(e) means for determining whether retransmitted lost frame packets have arrived prior to the frame's playout time;

(f) means for reconstructing said frame using the originally transmitted frame packets and, if necessary, said retransmitted lost frame packets, provided said retransmitted lost frame packets arrived before the expiration of the frame's playout time;

(g) display means for displaying the reconstructed frame;

(h) means for storing the just displayed reconstructed frame as said reference frame; and

(i) means for reconstructing said reference frame using the just displayed reconstructed frame and said retransmitted lost frame packets that did not arrive before the expiration of the frame's playout time.

17. The method of claim 13 further comprising the step of packetizing each frame into high priority signals and low priority signals wherein only high priority signal data is used for reconstruction of a frame.

18. The method of claim 12 further comprising the step of packetizing each frame into high priority signals and low priority signals wherein only high priority signal data is used for reconstruction of a frame.

19. The method of claim 14 further comprising the step of packetizing each frame into high priority signals and low priority signals wherein only high priority signal data is used for reconstruction of a frame.

21. A method of displaying an incoming stream of video images comprising the steps of:

(a) receiving said stream of video images in the form of frames comprised of packets, each frame based on a reference frame, and each frame having a

predesignated playout time;

(b) packetizing each frame into high priority signals and low priority signals wherein only high priority signal data is used for reconstruction of a frame;

(c) determining whether any frame packets were lost during transmission;

(d) requesting that any lost frame packets be retransmitted;

(e) determining whether retransmitted lost frame packets have arrived prior to the frame's playout time;

(f) reconstructing said frame using the originally transmitted frame packets and, if necessary, said retransmitted lost frame packets, provided said retransmitted lost frame packets arrived before the expiration of the frame's playout time;

(g) displaying the reconstructed frame;

(h) storing the just displayed reconstructed frame as said reference frame; and

(i) reconstructing said reference frame using the just displayed reconstructed frame and said retransmitted lost frame packets that did not arrive before the expiration of the frame's playout time.

22. A method of displaying an incoming stream of video images comprising the steps of:

(a) receiving said stream of video images in the form of frames comprised of packets, each frame based on a reference frame, and each frame having a predesignated extendable playout time wherein every i.sup.th frame has a variable extended temporal dependency distance that can be set to any number of frames while all other frames have a temporal dependency distance of one frame;

(b) packetizing each frame into high priority signals and low priority signals wherein only high priority signal data is used for reconstruction of a frame;

(c) determining whether any frame packets were lost during transmission;

(d) requesting that any lost frame packets be retransmitted;

(e) determining whether retransmitted lost frame packets have arrived prior to the frame's playout time;

(f) reconstructing said frame using the originally transmitted frame packets and, if necessary, said retransmitted lost frame packets, provided said retransmitted lost frame packets arrived before the expiration of the frame's playout time;

(g) displaying the reconstructed frame;

(h) storing the just displayed reconstructed frame as said reference frame; and

(i) reconstructing said reference frame using the just displayed reconstructed frame and said retransmitted lost frame packets that did not arrive before the expiration of the frame's playout time.

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L11: Entry 10 of 41 File: USPT Mar 19, 2002

DOCUMENT-IDENTIFIER: US 6360075 B1
TITLE: System and method for transmitting data

Application Filing Date (1):
19991001

Brief Summary Text (4):
Consumers have an insatiable appetite for information and entertainment, colloquially referred to as "content." This can be seen in the popularity of television, the internet and other content based media that are delivered to subscribers or users over various "pipelines." A pipeline is a system that transmits data from a content provider, e.g., television station, website on the internet, etc., to a subscriber. For example, internet service providers such as AmericaOn Line use the telephony system as a pipeline to transmit information to its subscribers. The subscribers use a computer modem to dial-in to an internet service provider. Once on-line, the subscribers have access to various content providers (websites) and can download or upload information. Unfortunately, this is often a slow and cumbersome technique for conveying large quantities of data because the telephony system has transmission speed and bandwidth limitations. Certain subscribers have installed specialized high-speed telephonic connections, but the practice is not widespread because of the prohibitive costs.

Brief Summary Text (5):
Similarly, various conventional pipelines deliver video information from content providers with varying degrees of success. Conventionally, television stations use a wireless pipeline for delivering content to users. The television stations simply broadcast signals in a dedicated portion of the electromagnetic spectrum. Users access the signals with roof-top antennas. Cable systems are also used in many areas. These systems use coaxial cable to deliver video with increased quality and quantity directly to a user's home or premises. However, conventional cable systems do not allow for interactive feedback to the content providers over the cable system. Retrofitting the existing cable systems with this feature will be expensive and time consuming.

Detailed Description Text (4):
Transmission system 10 can transmit audio, video, or other data for use by a computer, a television, a telephone or other appropriate terminal of subscribers 14. Transmission system 10 provides a pipeline between communication service providers 18 and subscribers 14. Communication service providers 18 may, for example, provide services such as video, interactive video, internet connection, telephony or access to other content based services. Transmission system 10 includes head end 17 coupled between communication service providers 18 and central hub 12. Head end 17 can communicate with communication service providers 18 and central hub 12 over any appropriate communication link such as wireless, including satellite and microwave or wired communication link as shown in FIG. 1. Transmission system 10 further includes a number of digital repeaters, represented here by digital repeaters 16a through 16c. It is understood that transmission system 10 includes an appropriate number of digital repeaters referred to collectively as "digital repeaters 16."

Detailed Description Text (7):
Transmission system 10 uses packets of digital data to increase the number of effective channels of the system. Specifically, transmission system 10 uses MPEG compression so as to transmit as many as 6 video channels in one 6 MHz channel. Similarly, transmission system 10 transmits multiple data channels in a single 6 MHz channel.

Detailed Description Text (18):
In one embodiment, transmission system 10 of FIG. 1 is controlled by a media access control (MAC) protocol. The MAC protocol supports video-on-demand services, data services, and control functions. The MAC protocol is an asymmetric protocol which uses fixed size MPEG 13818-1 transport stream transport packets for downstream transmissions and short, fixed size slots for upstream transmission. The protocol is consistent with DAVIC Technical Specifications 1.2 for MMDS by using transport packets for downstream. DAVIC presently leaves the upstream undefined ("reserved"). The MAC protocol of this embodiment is described in conjunction with FIGS. 5A through 10.

Detailed Description Text (21):
Downstream packet delivery is composed of two types of MPEG-2 transport packets, data transport packets and acknowledgment transport packets. Data transport packets are identified by a unique packet identification (PID) and acknowledgment transport packets are identified by a different unique PID. As illustrated in FIGS. 5A and 5B, the downstream data stream is composed of a sequence of 50 data transport packets 200 followed by an acknowledgment transport packet 202.

Detailed Description Text (24):
FIG. 6 is a diagram that illustrates the frame format of the downstream data packets 200. The basic form of the packet is an MPEG 13818-1 transport packet of fixed size (e.g., 188 bytes) and contains a 4 byte header. In general, transport packet PIDs are used to indicate program content including video and audio streams. In order to prevent any conflict with video programming the PID field is used only in the most limited manner. Specifically, two PIDs are used, one to indicate "normal" downstream data and the other for downstream acknowledgment.

Detailed Description Text (35):
6) The priority field is 1 bit that indicates the message priority to headend 17.

Detailed Description Text (68):
2. A single bit in the upstream packet could be used to distinguish user data from control data. A bit that is presently unused and could be used for this function is the priority bit.

Detailed Description Text (77):
Other timers may be required to implement non-essential components of the protocol, e.g., support for the priority scheme. These are not described here.

Detailed Description Text (78):
The protocol also provides a limited facility for expedited data upstream. The priority field in the upstream packet is used by the subscriber to indicate to headend 17 that the data in the packet should be expedited. This information may be used or ignored by headend 17 depending on the upstream algorithm implemented in headend 17.

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L6: Entry 5 of 5 File: USPT Feb 6, 2001

DOCUMENT-IDENTIFIER: US 6185602 B1
TITLE: Multi-user interaction of multimedia communication

Abstract Text (1):
The present invention provides multi-user interaction for multimedia communication. In one embodiment, a process for multi-user interaction for multimedia communication includes generating a message on a local user machine, the message including object-based media data (i.e., streamed, digital audio data or streamed digital video data or both), and transmitting the message to a remote user machine, in which the local user machine displays a scene that includes the object-based media data, the scene being shared by the local user machine and the remote user machine. The remote user machine constructs the message using a message handler class. In one embodiment, the multi-user inter-action for multimedia communication is an extension to MPEG-4 Version-1.

Brief Summary Text (11):
MPEG-4 represents an example of a media streaming technology for communicating digital multimedia over networks, such as the Internet (using the Internet Protocol), ATM (Asynchronous Transfer Mode) networks, mobile networks, or the PSTN (Public Switched Telephone Network). MPEG-4 (Version 1) is directed to a client-server architecture for object-based media broadcast in which a media server is generally assumed. However, MPEG-4 (Version 1) only supports single-user interaction. Accordingly, the present invention provides Multi-User Interaction (MUI) (i.e., at least two users interacting with each other dynamically, such as interacting with a shared scene, through servers or directly client-to-client) for multimedia communication. For example, the present invention provides a cost-effective and high-performance MUI protocol for MPEG-4 communication over the Internet. MUI for multimedia communication can be applied in a variety of application domains, such as collaborative computing, distance learning, shared virtual worlds, virtual chat rooms, entertainment, and E-commerce (Electronic-commerce), which involves interaction of two or more users with each other.

Brief Summary Text (12):
In one embodiment, a process for multi-user interaction for multimedia communication includes generating a message on a local user machine, the message including object-based media (i.e., streamed, digital audio or digital video or both) data, and transmitting the message to a remote user machine, in which the local user machine displays a scene that includes the object-based media data, the scene being shared by the local user machine and the remote user machine. The remote user machine instantiates the transmitted message using a message handler class. In one embodiment, the multi-user interaction for multimedia communication is an extension to MPEG-4 Version-1.

Detailed Description Text (14):
For example, MUI system 300 can be used to implement a virtual shopping mall. The virtual shopping mall can be represented by a scene graph, such as an MPEG-4 BIFS (Binary Format for Scene) scene, which is shared by a local machine and a remote machine. MPEG-4 BIFS provides a wrapping of object-based media by BIFS scene description as defined in MPEG-4 (Version-1), which allows for efficient intramedia, inter-media, and user-media interactions. Each shop in the virtual

shopping mall is a sub-scene. In one embodiment, each shop is implemented as a separate MUI system in which each MUI-system scene description is implemented locally without any reference to a remote MUI-system scene description. Accordingly, global extensibility and more efficient scene description are provided, and network access transparency and object access transparency between MUI systems are provided, as further discussed below.

Detailed Description Text (18):
In particular, state agent 504 extracts and maintains state information 508 of shared media objects. For example, in a shared environment, the local scene graph is modified if the state of another user's shared scene is modified. In this event, message agent 506 sends, creates, and receives arbitrary messages at runtime, including handling shared media object state changes. Message agent 506 sends messages 510 and 512. Messages 510 and 512 can include object-based media data, state information 508, control information, or any combination thereof.

Detailed Description Text (88):
FIG. 8 is a functional diagram of object-based multimedia messages transmitted across a network based on an MUI protocol for multimedia communication in accordance with one embodiment of the present invention. Local user machine 100 includes message class 702. Message class 702 instantiates a message object 802. Message object 802 includes control information and media data (i.e., streamed, digital audio data or streamed, digital video data or both). For example, message object 802 can include a new media object and control information regarding the location of the new media object in a shared scene (i.e., a scene shared by local machine 100 and remote (user) machine 202) Generally, message 802 can include control information, object-based media data, file data for an exchange of files between the local user and the remote user, confirmation data, status information, or any other data that supports an MUI environment. Message object 802 is transmitted to remote machine 202. The transmission can be a client-to-client transmission or a client-to-server (-to-client) transmission. In particular, message object 802 is wrapped in the appropriate protocol (e.g., DMIF and IP) and transmitted as a data signal over network 204 as indicated by a message 804. Remote machine 202 receives message 804. Message handler class 806 of remote machine 202 instantiates a message object 808, which includes the control information and new media object data of the transmitted message 802. For example, at this point, the MPEG-J APP of remote machine 202 can appropriately process the control information and new media object data to update the shared scene at remote machine 202 using the methods of the instantiated message object 808 as discussed above with respect to FIG. 7.

CLAIMS:

1. A process for multi-user interaction for multimedia communication, the process comprising:

generating a message in response to a change in a local scene displayed on a local user machine, the message comprising object-based media data; and

transmitting the message to a remote user machine to change a remote scene displayed on the remote user machine, the message containing information to modify the remote scene to reflect the change in the local scene using the object-based media data.

7. An article of manufacture for a computer-readable medium for multi-user interaction in multimedia communication, the article of manufacture comprising:

instructions for generating a message in response to a change in a local scene displayed on a local user machine, the message comprising object-based media data; and

instructions for transmitting the message to a remote user machine to change a remote scene displayed on the remote user machine, the message containing information to modify the remote scene to reflect the change in the local scene using the object-based media data.

14. The article of manufacture of claim 7 further comprising:

instructions for dynamically constructing a received message at the local user machine, wherein the received message was transmitted from the remote user machine, and the received message comprises object-based media data.

16. A machine for multi-user interaction for multimedia communication, the machine comprising:

a viewing device to display a local scene;

a message manager operatively coupled to the viewing device, the message handler configured to generate a message in response to a change in the local scene displayed on the viewing device, the message comprising object-based media data and information to modify a remote scene displayed at a remote user machine in accordance to the change in the local scene, the message manager further configured to transmit the message to the remote user machine; and

a microprocessor operatively coupled to the viewing device, the microprocessor configured to execute the message manager to implement the multi-user interaction.

21. An article of manufacture for a data signal in a carrier wave for multi-user interaction for multimedia communication, the data signal comprising:

an object-based media message, the object-based media message including information to modify a remote scene displayed at a remote machine in accordance to a change in a local scene displayed at a local machine, the object-based media message comprising object-based media data of the local scene; and

a network address of the remote machine,

wherein the object-based media message is transmitted to the remote machine using a message agent at the local machine.

22. The article of manufacture of claim 21 further comprising:

a control message, the control message comprising control information,

wherein the control message is transmitted to the remote machine, and the control information indicates location of the object-based media data in the remote scene, the local scene and the remote scene being a shared scene.

25. The article of manufacture of claim 24 wherein the object-based media message, the control message, and the state message each comprise a unique message object dynamically instantiated by a message class of the local machine, the message class comprising multiple messages of variable types.

27. The article of manufacture of claim 25 wherein the object-based media message is processed by the remote machine using a message handler class, the message handler class being implemented using a JAVA programming language.

First Hit Fwd Refs

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L13: Entry 3 of 10 File: USPT Jun 10, 2003

DOCUMENT-IDENTIFIER: US 6577596 B1
TITLE: Method and apparatus for packet delay reduction using scheduling and header compression

Abstract Text (1):
A method and apparatus for reducing delay in the transmission of a plurality of packets by performing IP scheduling and header compression at various layers in a multilayer architecture having a plurality of classifications includes scheduling packets according to classifications. If congestion occurs during scheduling, packets may be discarded. Some packet headers may be compressed thereafter. Packets may further be queued in a first and a second queue, after scheduling, discarding, and compressing, according to at least two classifications. Best Efforts packets may be queued into the lower priority second queue. Classifications may be associated with, for example, QoS levels, delay factors, LFI, and Multilink PPP. Scheduling is performed at higher or lower layers in a multi-layer protocol. The lower layer includes a PPP layer. The lower layer may also include an HDLC layer which creates a tag for packets prior to compression being performed thereupon. The tag may be added to packets at some point thereafter. Tags are removed prior to transmission. An outbound packet queue having a queue depth of no greater than one ensures no more than one Best Efforts packet wait time.

Application Filing Date (1):
19991130

Brief Summary Text (23):
It should be understood that a number of classifications may be included which may be associated with, for example, QoS levels as may be found in an IP header associated with a packet. Classifications may also be associated with a plurality of delay factors and it may be preferable to establish a plurality of queues based on the plurality of classifications. Accordingly, each packet may be queued into one of the plurality of queues based on an associated classification.

Detailed Description Text (5):
Notwithstanding LFI methods as described above, once packets are fragmented, they may be queued according to, for example, a conventional priority queuing scheme, an example of which is illustrated in FIG. 3, or may be queued according to a suitable derivative thereof. Exemplary network node 300 is shown having a priority queuing implementation with queues 311-314 ranging from Low to High priority respectively. Packets 315, 316, and 317, for example, arrive at network layer 210 with different priorities as may be determined by the contents of, for example, QoS information included with an IP header typically associated with each of packets 315, 316, and 317 respectively. High priority packet 317, for example, may be placed in high priority queue 314 by process 320. Incoming medium priority packet 315 and low priority packet 316 may be placed respectively in medium priority queue 313 and low priority queue 311 by process 320 when arriving at network layer 210. Priority queuing may take place at layer 310 which may be equivalent to data link layer 230 or an alternative protocol layer such as a PPP layer or the like which interfaces with data link layer 230. Outbound packets 318a-318d are sent to process 231 for queuing in transmit queue 232 according to priority with high priority outbound packet 318a being sent first as shown. Outbound packets 318a-318d may then be

transferred to the physical layer by process 231 which may be a typical data link process such as HDLC or the like where they can be processed in FIFO transmit queue 232 and output to physical link 233.

Detailed Description Text (8):
Therefore in accordance with one embodiment of the present invention, as illustrated in FIG. 5A, for example, various QoS levels which may be specified in an IP header of a datagram bound for IP layer 510 may be handled by performing "pure" IP scheduling at IP layer 510. It can be seen that a queuing discipline may be invoked using queues 512-514 for handling time sensitive packets. Lowest time sensitivity queue D.sub.N-1 512 my carry packets with the longest delay tolerance. It may be possible for such time sensitivity to be determined, for example, by examining the DS byte or ToS field associated with the typical IP header. Packets with lower QoS requirements, for example, may be relegated to lower priority queues. In contrast, packets with higher levels of QoS, such as real time packets associated with, for example voice data, may be associated with higher levels of time sensitivity and may accordingly be placed in higher priority queues. Packets with progressively greater degrees of time sensitivity may be scheduled in progressively higher priority queues with the highest sensitivity packets being scheduled to high time sensitivity queue D.sub.1 514. Packets having QoS set to Best Efforts, which are usually packets associated with non-real time data may be placed in Best Efforts queue D.sub.N 511, which as the name suggests, are sent when possible, for example, during intervals when there are no higher priority packets to be sent. It should be noted that interleaving and fragmentation may be used in conjunction with the time sensitive and Best Efforts queuing strategies as described. Processing in the pure IP scheduling embodiment will be described in greater detail hereinafter.

Detailed Description Text (12):
Before being transmitted on the physical line, a link layer adaptation is performed on an outgoing packet. In accordance with an embodiment performing scheduling at IP layer 510, HC and PPP/MP/HDLC framing may be performed to accomplish link layer adaptation. To shorten delay for time sensitive packets, such as voice packets, regardless of number of classes imposed by, for example, a fragmentation and interleaving scheme which may be used in accordance with the previous description, a simple priority queuing discipline may be used at PPP layer 520, using, for example, two queues, queue SQ_PPP 522 and queue FQ_PPP 523. If the degree of segmentation includes more than Best Efforts QoS or has additional classes as outlined in conjunction with multi-class PPP as described above, additional queues for each additional class to be segmented may be added at PPP layer 520. Accordingly, information related to when a packet can be transmitted must be known to the scheduling process, or in the present embodiment, PPP layer 520.

Detailed Description Text (13):
Packets classified by, for example, analyzing the DS or QoS field as described above, or by analyzing the class associated with the packet as outlined in reference to the description of FIG. 4A-4C, and grouped into delay levels associated, for example, with queues D.sub.1 514 to D.sub.N-1 512 from IP layer 510 may be placed in queue FQ_PPP 523, and will be sent before the rest of the segments belonging to, for example, a large BE packet. It should be noted that Header Compression may preferably be performed before packets are scheduled in queue FQ_PPP 523. BE packets and segmented packets may be put in queue SQ_PPP 522. If packets too large, MP processing in PPP layer 520 may perform segmentation before queuing.

CLAIMS:

1. A method of reducing delay in the transmission of a plurality of data packets on a communication link, said packets having a plurality of associated priority classifications, and being transmitted according to a multi-layer protocol having

at least a first layer and a second layer, the method comprising the steps of: scheduling each of the plurality of packets for transmission by placing each packet in one of a plurality of queues at the first layer, each of said first-layer queues corresponding to a different one of the plurality of priority classifications associated with the data packets; discarding any of the plurality of packets that cannot be placed in one of the first-layer queues; determining whether a queue having the highest priority in the second layer has sufficient space available for the packets in the highest priority first-layer queue to be moved to the highest priority second-layer queue; if the packets in the highest priority first-layer queue cannot be moved to the highest priority second-layer queue, retaining the packets in the highest priority first-layer queue until the packets can be moved to the highest priority second-layer queue; and if the packets in the highest priority first-layer queue can be moved to the highest priority second-layer queue: compressing a header on each of the packets in the highest priority first-layer queue; moving the packets in the highest priority first-layer queue to the highest priority second-layer queue; and transmitting the packets from the highest priority second-layer queue through an output buffer to the communication link.

2. The method of claim 1, wherein the plurality of classifications are associated with Quality of Service (QoS) levels.

13. An apparatus for reducing delay in the transmission of a plurality of data packets on a communication link, said packets having a plurality of associated priority classifications, said apparatus comprising: a plurality of prioritized queues at a first layer of the apparatus, each queue having a priority corresponding to one of the plurality of priority classifications, said priorities ranging from a highest priority queue to a lowest priority queue; a high priority queue and a low priority queue at a second layer of the apparatus; a scheduling function that places each of the plurality of data packets in a first-layer queue having a priority that corresponds to the priority classification of the packet; a discarding function that discards any of the plurality of packets that cannot be placed in one of the first-layer queues; a packet moving and compressing function that determines whether the high priority second-layer queue has sufficient space available for the packets in the highest priority first-layer queue to be moved to the high priority second-layer queue, and if so, compresses a header on each of the packets in the highest priority first-layer queue, and moves the packets in the highest priority first-layer queue to the high priority second-layer queue; an output buffer between the communication link and the second-layer queues; and a transmission function that transmits the packets from the high priority second-layer queue through the output buffer to the communication link.

14. The apparatus of claim 13, wherein the plurality of classifications are associated with different Quality of Service (QoS) levels.

L13: Entry 5 of 10 File: USPT Apr 15, 2003

DOCUMENT-IDENTIFIER: US 6549938 B1
TITLE: System and method for prioritizing multicast packets in a network service class utilizing a priority-based quality of service

Abstract Text (1):
A system and method for achieving a comparable quality of service for each of the receivers of a multicast transmission incorporating a priority-based quality of service is provided. Packet acceptance criteria established at each individual connection of a network node is overridden to provide a collective packet acceptance criteria for each packet of a multicast transmission targeted for the individual connections. The packet acceptance criteria is collected from each of the individual connections in the network node that are targeted for the multicast transmission. A multicast packet priority is calculated for each of the packets associated with the multicast transmission based on an aggregate analysis of the packet acceptance criteria of each of the individual connections. Each of the packets associated with the multicast transmission is collectively accepted or discarded based on the calculated multicast packet priority.

Application Filing Date (1):
19981210

Brief Summary Text (8):
A conventional ATM service architecture typically provides a number of predefined quality of service classes, often referred to as service categories. Each of the service categories includes a number of quality of service (QoS) parameters which define the nature of the respective service category. In other words, a specified service category provides performance to an ATM virtual connection (VCC or VPC) in a manner specified by a subset of the ATM performance parameters. The service categories defined in the ATM Forum specification reference hereinbelow include, for example, a constant bit rate (CBR) category, a real-time variable bit rate (rt-VBR) category, a non-real-time variable bit rate (nrt-VBR) category, an unspecified bit rate (UBR) category, and an available bit rate (ABR) category.

Brief Summary Text (9):
The constant bit rate service class is intended to support real-time applications that require a fixed quantity of bandwidth during the existence of the connection. A particular quality of service is negotiated to provide the CBR service, where the QoS parameters include characterization of the peak cell rate (PCR), the cell loss rate (CLR), the cell transfer delay (CTD), and the cell delay variation (CDV). Conventional ATM traffic management schemes guarantee that the user-contracted QoS is maintained in order to support, for example, real-time applications, such as circuit emulation and voice/video applications, which require tightly constrained delay variations.

Brief Summary Text (17):
Multicasting is the transmission of packets from one source to multiple receivers or users. For example, in a video broadcasting application, the server sends the same picture to every client. A problem is that that a priority-based QoS such as implemented in SIMA can lead to quite different quality of service at each of the different branches of the multicast transmission. While this may be acceptable in

many situations, there are situations where it is preferable to guarantee a similar quality of service to all receivers of the multicast transmission. For example, a server providing video multicast may want to ensure that all of its clients receive a video picture that has an approximately constant quality regardless of the location of the receiving user.

Brief Summary Text (21):
In accordance with one embodiment of the invention, a method is provided for overriding packet acceptance criteria established at each individual connection of a network node, to provide a collective packet acceptance criteria for each packet of a multicast transmission targeted for the individual connections. The packet acceptance criteria is collected from each of the individual connections in the network node that are targeted for the multicast transmission. A multicast packet priority is calculated for each of the packets associated with the multicast transmission based on an aggregate analysis of the packet acceptance criteria of each of the individual connections. Each of the packets associated with the multicast transmission is collectively accepted or discarded based on the calculated multicast packet priority.

Brief Summary Text (22):
In accordance with another embodiment of the invention, a method is provided for overriding packet acceptance criteria established at each individual connection of a network node, to provide a collective packet acceptance criteria for each packet of a multicast transmission targeted for the individual connections. The packet acceptance criteria is collected from each of the individual connections in the network node that are targeted for the multicast transmission. A multicast packet priority is calculated for each of the packets associated with the multicast transmission based on an aggregate analysis of the packet acceptance criteria of each of the individual connections. The packet acceptance criteria from each of the connections in the network node is modified based on the calculated multicast packet priority. Each of the packets associated with the multicast transmission is individually accepted or discarded based on the modified packet acceptance criteria.

CLAIMS:

1. A method for overriding packet acceptance criteria established at each individual connection of a network node, to provide a collective packet acceptance criteria for each packet of a multicast transmission targeted for the individual connections, comprising: collecting the packet acceptance criteria from each of the individual connections in the network node that are targeted for the multicast transmission; calculating a multicast packet priority for each of the packets associated with the multicast transmission based on an aggregate analysis of the packet acceptance criteria of each of the individual connections; collectively accepting or discarding each of the packets associated with the multicast transmission based on the calculated multicast packet priority; and determining whether the packet is a packet associated with a multicast transmission, and bypassing the collecting of the packet acceptance criteria, bypassing the calculating of a multicast packet priority, and bypassing the collective accepting or discarding of each of the packets when the packet is not associated with the multicast transmission.

6. A method for overriding packet acceptance criteria established at each individual connection of a network node, to provide a collective packet acceptance criteria for each packet of a multicast transmission targeted for the individual connections, comprising: collecting the packet acceptance criteria from each of the individual connections in the network node that are targeted for the multicast transmission; calculating a multicast packet priority for each of the packets associated with the multicast transmission based on an aggregate analysis of the packet acceptance criteria of each of the individual connections; collectively

accepting or discarding each of the packets associated with the multicast transmission based on the calculated multicast packet priority; and determining whether the packet is a packet associated with a multicast transmission, and bypassing the collecting of the packet acceptance criteria, bypassing the calculating of a multicast packet priority, and bypassing the collective accepting or discarding of each of the packets when the packet is not associated with the multicast transmission, wherein bypassing further comprises: using the packet acceptance criteria established at each individual connection of the network node as a basis to individually accept or reject the packets not associated with the multicast transmission.

10. A method for overriding packet acceptance criteria established at each individual connection of a network node, to provide a collective packet acceptance criteria for each packet of a multicast transmission targeted for the individual connections, comprising: collecting the packet acceptance criteria from each of the individual connections in the network node that are targeted for the multicast transmission, wherein collecting the packet acceptance criteria from each of the individual connections comprises: receiving an allowable packet priority corresponding to each of the individual connections, wherein the allowable packet priority represents a minimum packet priority necessary for acceptance; calculating a multicast packet priority for each of the packets associated with the multicast transmission based on an aggregate analysis of the packet acceptance criteria of each of the individual connections; modifying the packet acceptance criteria at each of the individual connections in the network node based on the calculated multicast packet priority, wherein modifying the packet acceptance criteria comprises: increasing the allowable packet priorities of each of the individual connections to increase a likelihood of packet acceptance at each of the individual connections; and individually accepting or discarding each of the packets associated with the multicast transmission based on the modified packet acceptance criteria.

13. A method for overriding packet acceptance criteria established at each individual connection of a network node, to provide a collective packet acceptance criteria for each packet of a multicast transmission targeted for the individual connections, comprising: collecting the packet acceptance criteria from each of the individual connections in the network node that are targeted for the multicast transmission, wherein collecting the packet acceptance criteria from each of the individual connections comprises: receiving an allowable packet priority corresponding to each of the individual connections, wherein the allowable packet priority represents a minimum packet priority necessary for acceptance; calculating a multicast packet priority for each of the packets associated with the multicast transmission based on an aggregate analysis of the packet acceptance criteria of each of the individual connections; modifying the packet acceptance criteria at each of the individual connections in the network node based on the calculated multicast packet priority, wherein modifying the packet acceptance criteria comprises: decreasing the allowable packet priorities of each of the individual connections to decrease a likelihood of packet acceptance at each of the individual connections; and individually accepting or discarding each of the packets associated with the multicast transmission based on the modified packet acceptance criteria.

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L13: Entry 6 of 10 File: USPT Feb 25, 2003

DOCUMENT-IDENTIFIER: US 6526062 B1
TITLE: System and method for scheduling and rescheduling the transmission of cell objects of different traffic types

Abstract Text (1):
A computer system for transmitting packets includes a manager and scheduling elements for managing the transmission of the packets over one or more logical channels. The computer system can prioritize the transmission of packets based on the type of traffic and maintain quality of service (QoS) characteristics associated with a logical channel. In addition, the computer system can execute a threading process to ensure the efficient and timely transmission of certain types of packets without using any complex mathematical operations.

Application Filing Date (1):
19981013

Brief Summary Text (5):
With the advent of highly-sophisticated digital audio and video applications run on modern multimedia workstations, the ability to run these applications over the Internet has become more desirable. However, such real-time applications often do not work well across the Internet because of variable queuing delays and congestion losses, and because the Internet, as conceived, offers only a very simple point-to-point, best-effort data delivery. As a result, before real-time applications, such as remote video and multimedia conferencing, can be broadly used, the Internet infrastructure must be modified to support real-time quality of service (QoS), which provides some control over end-to-end packet delays.

Brief Summary Text (6):
Another problem with respect to communication over the Internet involves the communication lines. Long haul communications lines are very expensive to use, and major customers usually contract to pay for the use of these lines according to the amount of "time" they wish to have access to these lines rather than by the amount of traffic they send over them. Consequently, it is very important that these customers make the most efficient use of these lines. To make efficient use of these lines, it is desirable to provide requested/specified asynchronous transfer mode (ATM) QoS for many thousands of virtual channels (VCs) all using the same physical port. In other words, a network should be optimally setup so that traffic meets the specified QoS parameters of any portion of the network.

Brief Summary Text (9):
With respect to the Internet protocol (IP) QoS problem, prior art solutions have implemented multiple logical FIFOs to handle variously prioritized packets, typically referred to as "priority queuing." The queue with the highest priority traffic would always be checked first for an available packet to send and when this queue was emptied, the next priority queue would be checked for available packets to send, and so forth. Such a priority queuing arrangement, however, does not guarantee service to every packet because high priority traffic can "lock out" low priority traffic indefinitely. In contrast, by giving, for example, all users in a packet-scheduling scheme the same priority, but treating the queues in a round robin fashion, packet scheduling guarantees each user a particular committed amount

of bandwidth with which other users cannot interfere. Alternatively, each user can have a variable priority based on whether the user has been under or over-utilizing their guaranteed bandwidth. The weighted fair queuing (WFQ) algorithm provides such a scheme.

Detailed Description Text (6):
PQM 230 is composed of two queues, a packet linked list queue manager (PLLQM) 240 and a packet sorted tree queue manager (PSTQM) 250. Each queue can be implemented, for example, as a field programmable gate array (FPGA). Other implementations include application specific integrated circuits (ASICs). PLLQM 240 receives packet objects from PSAP 210 that are associated with "Best Effort" traffic and normal cell bridging traffic, while PSTQM 250 receives packet objects from PSAP 210 that are associated with QoS traffic and certain operation and maintenance functions. Based on information generated by PSAP 210 and exhaustion signals received from cell scheduler 145, PSTQM 250 and PLLQM 240 operate to enqueue and dequeue packet objects into and out of a packet queue memory 280 via address/WR data lines. PLLQM 240 and PSTQM 250 share packet queue memory 280 through a multiplexer 270.

Detailed Description Text (31):
The VBR scheduling calculation process as described above works well as long as the LCI capacity is not over allocated. The primary function of CSCHEDS 330 and 340, as stated earlier, is to calculate the earliest acceptable time that a cell can be sent out (QT). This time is placed in the VBR TQ as an index, and when CT moves to that time slot in the VBR TQ, the cell would be transmitted or, in the worst case, moved to the next slot. The CSCHEDS 330 and 340 calculate QT using several internally stored values, which includes Is, Ip, and Is-L. These values can be different for each LCI depending upon the QoS characteristics selected for the LCI. These values are used by CSCHEDS 330 and 340 to calculate other internally stored values, such as TDT and SDT, which are in turn used by the algorithm to calculate the QT, as describe above.

CLAIMS:

1. A method for scheduling the transmission of packet objects, each packet object having at least one cell and associated with a logical channel identifier (LCI) identifying the logical channel over which the packet object is being transmitted, comprising the steps of: determining whether an LCI is scheduled to be transmitted at a current time; transmitting at the current time a cell of a packet object corresponding to the LCI scheduled for transmission at the current time according to a priority associated with a traffic type of the LCI, at least one of the traffic types being a predetermined traffic type; and rescheduling an LCI of the predetermined traffic type to a queue time (QT), which corresponds to a transmission time later than the current time, when at least one cell remains for the LCI of the predetermined traffic type scheduled for transmission at the current time wherein the rescheduling step comprises: determining a scheduled departure time (SDT) by calculating a first time as the sum of a theoretical departure time (TDT) and a first increment, calculating a second time as the sum of a previously calculated SDT and a second increment, and selecting one of the first and second times having the later time as the determined SDT; and setting the QT as the later time of the determined SDT and the transmission time immediately after the current time, wherein the next theoretical departure time is calculated as the sum of the prior value of TDT and a third increment.

8. A method for scheduling the transmission of packet objects, each packet object having at least one cell and associated with a logical channel identifier (LCI) identifying the logical channel over which the packet object is being transmitted, comprising the steps of: determining whether an LCI is scheduled to be transmitted at a current time; transmitting at the current time a cell of a packet object corresponding to the LCI scheduled for transmission at the current time according to a priority associated with a traffic type of the LCI, at least one of the

traffic types being a predetermined traffic type; rescheduling an LCI of the predetermined traffic type to a queue time (QT), which corresponds to a transmission time later than the current time, when at least one cell remains for the LCI of the predetermined traffic type scheduled for transmission at the current time; receiving a new packet object of the predetermined traffic type; determining a scheduled departure time (SDI) for the LCI of the new packet object by calculating a first time as the sum of a theoretical departure time (TDT) and a first increment, calculating a second time as the sum of a previously calculated SDT and a second increment, and selecting one of the first time, the second time, and a next time, which corresponds to a transmission time immediately after the current time, having the later time as the determined SDT; and setting the QT of the LCI of the new packet object as the determined SDT, wherein the next theoretical departure time is determined by calculating a third time as the sum of the prior value of TDT and a third increment and selecting one of the first time and the next time having the later time as the next theoretical departure time.

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L13: Entry 8 of 10 File: USPT Mar 20, 2001

DOCUMENT-IDENTIFIER: US 6205150 B1
TITLE: Method of scheduling higher and lower priority data packets

Application Filing Date (1):
19980528

Brief Summary Text (2):
The present invention relates to communications in computer networks. More specifically, it relates to a method for dynamically scheduling transmission of high and low priority data packets associated with a network device by utilizing dual queues, dual scheduling methods and a promoter.

Brief Summary Text (9):
The problems associated with scheduling tasks in an operating system also occur in multi-user network systems with a plurality of network connections, network devices and data packets. In a network system environment, a data packet is analogous to a task in an operating system. Customers on a network system may have different Customer Premise Equipment ("CPE") (i.e., a computer) with different capabilities, such as the ability to send and receive data packets at various data rates or bandwidth. In a multimedia system, logical multimedia channels are typically used by a network connection to create separate audio, video and data channels. The audio and video channels are typically allocated with predetermined, fixed maximum bandwidth. For example, on a modem connection an audio channel may have a bandwidth of 5,300 bits-per-second (bps) and a video channel may have a bandwidth of 23,500 bps for a multimedia bandwidth of 28,800 bps (i.e., the sum of the two channels). Many network hosts allow customers to subscribe to various Classes-of-Service ("CoS") and Qualities-of-Service (`QoS") to optimize reliability and data transmission speeds. As is known in the art, class-of-service provides a reliable (e.g., error free, in sequence, with no loss of duplication) transport facility independent of the quality-of-service. Class-of-service parameters include maximum downstream data rates, maximum upstream data rates, upstream channel priority, guaranteed minimum data rates, guaranteed maximum data rate and others. Quality-of-service collectively specifies the performance of a network service that a device expects on a network. Quality-of-service parameters include transit delay expected to deliver data to a specific destination, the level of protection from unauthorized monitoring or modification of data, cost for delivery of data, expected residual error probability, the relative priority associated with the data and other parameters. Higher class-of-service and quality-of-service connections transmit higher priority data packets. Thus, various customers on the network system will transmit and receive both high priority and low priority data packets.

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L14: Entry 22 of 44 File: USPT Sep 26, 2000

DOCUMENT-IDENTIFIER: US 6125110 A
TITLE: Method and system for determining a packet transmission order

Application Filing Date (1):
19980731

Detailed Description Text (14):
In the preferred embodiment, NAs 50, 52 and 54 receive speech packets from a plurality of selectors, corresponding to a plurality of mobile units. NAs 50, 52 and 54 utilize the priority level associated with each packet to determine the order of transmission of the packets into communication links 17, 19 and 21. For example, speech packets destined for mobile unit 22b must be simultaneously transmitted over the air interface by BSs 18 and 20. However, communication link 21 is longer than communication link 19 and hence the packet transmission time from NA 54 to NA 84 is greater than that from NA 52 to NA 82. Hence, a speech packet destined for mobile unit 22b can be buffered longer at arbiter 52 while still ensuring simultaneous transmission via the air interface. Similarly, speech packets destined for mobile unit 22a must be simultaneously transmitted over the air interface by BSs 16 and 18. The packet transmission time from NA 52 to NA 82 is greater than that from NA 50 to NA 80. Hence, speech packets destined for mobile unit 22a can be buffered longer at arbiter 50 than arbiter 52.

Detailed Description Text (30):
Communication 2 is a one-way call, not in soft handoff. Following the priority scheme of the preferred embodiment, packets associated with communication 2 are assigned a priority [2] for transmission to BS 122.

Detailed Description Text (32):
Communication 4 is a one-way communication, not in soft handoff. Following the priority scheme of the preferred embodiment, packets associated with communication 4 are assigned a priority [2] for transmission to BS 126.

CLAIMS:

11. A method for determining the order of transmission of packets of information, comprising the steps of:

receiving a plurality of packets of information, each packet being associated with differnt mobile units and at least a first base station;

determining for each packet whether the associated communication is between the associated mobile unit and either the first base station, the associated mobile unit and two base stations or the associated mobile unit and three base stations;

assigning a first priority level with those packets associated with mobile unit only in communication with the first base station;

assigning a second priority level with those packets associated with mobile units in communication with two base stations when a transmission delay associated with the first base station is greater than a transmission delay associated with a

second base station;

assigning a third priority level with those packets associated with mobile units in communication with two base stations when a transmission delay associated with the first base station is less than a transmission delay associated with the second base station; and

sending those packets assigned with a second priority level before those packets assigned with a third priority level.

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L14: Entry 2 of 44 File: PGPB Aug 9, 2001

DOCUMENT-IDENTIFIER: US 20010012293 A1
TITLE: SIMULTANEOUS TRANSMISSION OF VOICE AND NON-VOICE DATA ON A SINGLE NARROWBAND CONNECTION

Application Filing Date:
19971202

Detail Description Paragraph:
[0043] The control logic unit 550 and the minicell assembly module 545 work in conjunction with each other to transform the data packets, stored in the FIFO 540, into minicells and to multiplex those minicells into a single data stream. Upon receiving a data packet from one of the voice and/or non-voice sources, the FIFO 540 sends a control signal to the control logic unit 550. The control logic unit 550, in turn, commands the minicell assembly module 545 to select the data packet, thereby initiating the process of transforming the data packet into a minicell format. However, if there is more than one data packet stored in the FIFO 540, the control logic unit 550 commands the minicell assembly module 545 to select the data packets in accordance with a predefined priority scheme. Generally, data packets associated with voice data sources are assigned a higher priority than data packets associated with non-voice sources, as voice data is highly sensitive to transmission delays.

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L14: Entry 21 of 44 File: USPT Nov 14, 2000

DOCUMENT-IDENTIFIER: US 6147980 A
TITLE: Avionics satellite based data message routing and delivery system

Application Filing Date (1):
19971128

CLAIMS:

3. The avionics data message routing and delivery system of claim 2, wherein:

said communications controller transmits said data packets temporarily stored in said transmission queue in order of a data packet priority associated with each of said data packets.

4. The avionics data message routing and delivery system of claim 3, said controller further comprising:

a transmission prioritizing controller for determining said data packet priority associated with each of said data packets temporarily stored in said transmission queue.

5. The avionics data message routing and delivery system of claim 4, wherein:

said transmission prioritizing controller determines said data packet priority associated with each of said data packets temporarily stored in said transmission queue based on priority bits encoded in each of said data packets temporarily stored in said transmission queue.

19. The avionics data message routing and delivery system of claim 18, wherein:

said communications controller transmits said data packets temporarily stored in said transmission queue in order of a data packet priority associated with each of said data packets.

20. The avionics data message routing and delivery system of claim 19, said controller further comprising:

a transmission prioritizing controller for determining said data packet priority associated with each of said data packets temporarily stored in said transmission queue.

21. The avionics data message routing and delivery system of claim 20, wherein:

said transmission prioritizing controller determines said data packet priority associated with each of said data packets temporarily stored in said transmission queue based on priority bits encoded in each of said data packets temporarily stored in said transmission queue.

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L14: Entry 3 of 44 File: USPT Oct 14, 2003

DOCUMENT-IDENTIFIER: US 6633564 B1
TITLE: Method and apparatus for inserting packets into a data stream

Application Filing Date (1):
19990922

CLAIMS:

37. The scheduler of claim 36 wherein the scheduler receives packets from a plurality of queues, each queue being associated with a particular transmission priority.

38. The scheduler of claim 37 wherein for determining whether to treat a packet received as an interrupting packet, the scheduler relies on the transmission priority associated with a queue containing the packet.

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L14: Entry 20 of 44 File: USPT Feb 13, 2001

DOCUMENT-IDENTIFIER: US 6188670 B1
TITLE: Method and system in a data processing system for dynamically controlling transmission of data over a network for end-to-end device flow control

Application Filing Date (1):
19971031

Brief Summary Text (3):
The present invention relates to data processing systems and, in particular, to dynamically controlling transmission of data over a network within a data processing system. Still more particularly, the present invention relates to a method and system in a data processing system for permitting a receiver to dynamically alter a transmission priority level associated with multiple real-time data packets during transmission of real-time and non-real-time packets over a network in order to provide end-to-end flow control between a sender and a receiver attached to the network.

Brief Summary Text (20):
It is yet another object of the present invention to provide a method and system in a data processing system for permitting a receiver to dynamically alter a transmission priority level associated with multiple real-time data packets during transmission of real-time and non-real-time packets over a network.

Detailed Description Text (5):
When these files are presented, they will need to be transmitted utilizing a network from a sender to a receiver. For example, a video file which is stored in a storage device on one computer may need to be displayed on the display screen of another computer in the network. In this case, the file must be transmitted from the storage device to be displayed on the screen. The file is transmitted utilizing the network. Also during this time, non-real-time data may need to be transmitted utilizing the same network. For example, file transfers may need to be performed. Both real-time data and non-real-time data may need to be transmitted simultaneously utilizing the same network. The present invention provides a method and system for permitting a receiver to dynamically alter a transmission priority level associated with multiple real-time data packets during transmission of real-time and non-real-time packets over a network.

Detailed Description Text (19):
Real-time data may be associated with one of a large number of different priorities. These priorities may be grouped and associated with one of a multiple of transmission priority levels. For example, there may be twenty-five different real-time priorities. At the time of transmission, a packet of real-time data may be associated with one of these twenty-five different priorities. For purposes of illustration only, the first ten priorities may be grouped and then associated with a higher transmission priority level. The remaining fifteen priorities may be grouped and associated with a lower transmission priority level. Those skilled in the art will recognize that the priorities may be grouped into any number of groups. For example, there may be three groups, each associated with one of three different transmission priority levels.

First Hit Fwd Refs

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L14: Entry 27 of 44 File: USPT Sep 1, 1998

DOCUMENT-IDENTIFIER: US 5802051 A
TITLE: Multiplexing of voice and data minicells

Application Filing Date (1):
19960610

Brief Summary Text (19):
In accordance with one aspect of the invention, the foregoing and other objects are achieved by a method and/or an apparatus for multiplexing segmented user data packets into a stream of data cells prior to transmission from a sending station to a receiving station, the method comprising the steps of: dividing a user data packet into segments; assigning a transmission priority code to each segment based on a type of data associated with the user data packet; and multiplexing the segments into the stream of data cells with at least one segment from another user data packet as a function of transmission priority, wherein said segments are interleaved relative to one another as a function of transmission priority.

Brief Summary Text (20):
In accordance with another aspect of the invention, the foregoing and other objects are achieved by a method and/or an apparatus for multiplexing segmented user data packets into a stream of data cells prior to transmission from a sending station to a receiving station, the method comprising the steps of: retrieving a user data packet from one of a select number of telecommunication applications serviced by said telecommunication system, wherein said telecommunication application generates a plurality of different data types; dividing the user data packet into segments according to the length of said user data packet and the type of data associated with said user data packet; providing a corresponding minicell for each segment of the user data packet; assembling each of the segments of said user data packet into a payload portion of the corresponding minicell; attaching to each minicell, a minicell header containing a code that identifies a transmission priority associated with a type of data contained in the segment; and multiplexing the minicells into a stream of data cells with minicells associated with other user data packets from the same telecommunication application for transmission to a receiving station, wherein the order in which the minicells are multiplexed is a function of transmission priority.

Detailed Description Text (9):
FIGS. 7A and 7B further illustrate how the length codes in the header of each segment minicell are used to define not only the length and relative position of each segment minicell, but also the transmission priority of the corresponding user data packet. Referring first to FIG. 7A, the length code 52 or 53 indicates that a) the segment minicell is a first or middle segment minicell, b) the segment minicell is 16 octets or 32 octets in length respectively; and c) the segment minicell is associated with a user data packet belonging to transmission priority category "2", circuit data. Referring next to FIG. 7B, the length code 54 or 55 indicates that a) the segment minicell is a last segment minicell, b) the segment minicell is 8 octets or 16 octets in length respectively; and c) the segment minicell is associated with a user data packet belonging to transmission priority category "2", circuit data.

Detailed Description Text (10):
FIG. 7B also illustrates another aspect of the invention in which a somewhat different protocol is employed to define the last segment of user data packets associated with transmission priority categories "2" through "5" than is utilized in prior art methods. More specifically, FIG. 7B illustrates that a last segment minicell associated with transmission priority category "2" is limited to a fixed length of either 8 or 16 octets. Similarly, the last segment minicells associated with the other transmission priority categories other than category "1" are also limited to one or two fixed lengths. By contrast, in prior art methods, the last segment minicell could take on any length, essentially limited only by the length of the ATM cell payload. Because it is desirable for the inventive segmenting technique to be compatible with prior art techniques with respect to providing category "1"-type data packets, a preferred embodiment of the invention allows the length of a last segment minicell associated with transmission priority category "1" to vary from just 1 octet to 46 octets.

Detailed Description Text (12):
Given a 6 bit length field and the limited number of available length code combinations, as explained above, the present invention provides a modified user data packet segmentation process for user data packets associated with transmission priority categories "2" through "5". This modified segmentation process is described hereinbelow.

Detailed Description Text (13):
FIG. 8 illustrates an exemplary segmentation method 800 to be executed, in accordance with the present invention, by the SAR sublayer 801 and the AAD sublayer 805 on a user data packet 810, which is associated with transmission priority category "2", "3", or "4". After the user data packet 810 arrives at the SAR sublayer 801, the SAR sublayer 801 extends the length by a certain number of octets. The extension is depicted by the trailer 815.

CLAIMS:

1. In a telecommunication system, a method of multiplexing user data packets into a data stream prior to transmission from a sending station to a receiving station, the method comprising the steps of:

dividing a first user data packet into a first plurality of segments if the first user data packet is greater in length than a first predefined length, wherein the first user data packet has associated with it a first data type and a first transmission priority based on the first data type;

dividing a second user data packet into a second plurality of segments if the second user data packet is greater in length than the first predefined length, wherein the second user data packet has associated with it a second data type and a second transmission priority based on the second data type;

generating a minicell header for each segment associated with the first plurality of segments, wherein each minicell header contains a code which identifies the first data type, the first transmission priority, and a position of the corresponding segment in the first user data packet;

generating a minicell header for each segment associated with the second plurality of segments, wherein each minicell header contains a code which identifies the second data type, the second transmission priority, and a position of the corresponding segment in the second user data packet;

appending each minicell header to its corresponding segment to form a first plurality of minicells associated with the first user data packet and a second plurality of minicells associated with the second user data packet; and

multiplexing the minicells associated with the first plurality of minicells and the minicells associated with the second plurality of minicells into the data stream, wherein the minicells associated with the first plurality of minicells and the minicells associated with the second plurality of minicells are interleaved relative to one another as a function of transmission priority.

5. In a telecommunication system, a method of multiplexing user data packets into a stream of data cells prior to transmission from a sending station to a receiving station, the method comprising the steps of:

retrieving a first user data packet from one of a select number of telecommunication applications serviced by said telecommunication system, wherein said telecommunication application generates a plurality of different data types;

dividing the first user data packet into segments according to the length and data type of the first user data packet;

generating a minicell corresponding to each segment by attaching a minicell header to each segment, wherein each minicell header contains a code that identifies the data type, a transmission priority associated with the data type, a length of the corresponding segment, and a position of the corresponding segment in the first user data packet; and

multiplexing the minicells into the stream of data cells, wherein the stream of data cells contains minicells <u>associated with a second user data packet, and wherein the order in which the minicells associated with the first user data packet and the second user data packets are multiplexed is a function of transmission priority</u>.

11. In a telecommunication system, an apparatus for multiplexing user data packets into a data stream prior to transmission from a sending station to a receiving station, the apparatus comprising:

first segmentation means for dividing a first user data packet into a first plurality of segments if the first user data packet is greater in length than a first predefined length, wherein the first user data <u>packet has associated with it a first data type and a first transmission priority</u> based on the first data type;

second segmentation means for dividing a second user data packet into a second plurality of segments if the second user data packet is greater in length than the first predefined length, wherein the second user data <u>packet has associated with it a second data type and a second transmission priority</u> based on the second data type;

means for generating a minicell header for each segment associated with the first plurality of segments, wherein each minicell header contains a code which identifies the first data type, the first transmission priority, and a position of the corresponding segment in the first user data packet;

means for generating a minicell header for each segment associated with the second plurality of segments, wherein each minicell header contains a code which identifies the second data type, the second transmission priority, and a position of the corresponding segment in the second user data packet;

assembly means for appending each minicell header to its corresponding segment to form a first plurality of minicells associated with the first user data packet and a second plurality of minicells associated with the second user data packet; and

multiplexing means for inserting the minicells associated with the first plurality

of minicells and the minicells associated with the second plurality of minicells into the data stream, wherein said minicells associated with the first plurality of minicells and the minicells associated with the second plurality of minicells are interleaved relative to one another as a function of transmission priority.

15. In a telecommunication system, an apparatus for multiplexing user data packets into a stream of data cells prior to transmission from a sending station to a receiving station, the apparatus comprising:

means for retrieving a first user data packet from one of a select number of telecommunication applications serviced by said telecommunication system, wherein said telecommunication application generates a plurality of different data types;

segmentation means for dividing the first user data packet into segments according to the length and data type of the first user data packet;

assembly means for generating a minicell corresponding to each segment by attaching a minicell header to each segment, wherein each minicell header contains a code that identifies the data type, a transmission priority associated with the data type, a length of the corresponding segment, and a position of the corresponding segment in the first user data packet; and

multiplexing means for inserting the minicells into the stream of data cells, wherein the stream of data cells contains minicells associated with a second user data packet,

wherein the order in which the minicells associated with the first user data packet and the second user data packet are multiplexed is a function of transmission priority.

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L14: Entry 14 of 44 File: USPT Nov 20, 2001

DOCUMENT-IDENTIFIER: US 6320845 B1
TITLE: Traffic management and flow prioritization on a routed computer network

Application Filing Date (1):
19980427

CLAIMS:

4. The apparatus of claim 1 wherein the selection criterion is determined by a weighted fair queueing scheme, the prioritization means associating different packets with different priority levels and selecting packets for transmission in accordance with the priority levels.

10. A routing device configured for management of packet flows, the device comprising:

a. at least one input port for receiving flows;

b. at least two output ports for transmitting flows, each port having a maximum rate of packet transmission, the maximum rates varying among the at least two output ports; and

c. prioritization means for (i) queueing packets arriving over an input port for transmission over a designated output port and (ii) selecting queued packets for transmission based on a selection criterion, the selection criterion being determined by a weighted fair queueing scheme, the prioritization means associating different packets with different priority levels and selecting packets for transmission in accordance with the priority levels, the prioritization means interacting with the ports and postponing transmission of selected packets where such transmission would exceed the maximum rate of the designated output port, the prioritization means postponing selection of a packet by an amount of time proportional to the priority level associated with the packet.

17. The network interface of claim 13 wherein the selection criterion is determined by a weighted fair queueing scheme, the prioritization means associating different packets with different priority levels and selecting packets for transmission in accordance with the priority levels.

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L13: Entry 2 of 10 File: USPT Dec 9, 2003

DOCUMENT-IDENTIFIER: US 6661774 B1
TITLE: System and method for traffic shaping packet-based signals

Application Filing Date (1):
19990216

Brief Summary Text (9):
In accordance with the present invention, a system and method for traffic shaping packet-based signals are provided that substantially eliminate or reduce disadvantages or problems associated with previously developed systems and methods. In particular, the present invention facilitates traffic shaping signal packets to provide an efficient distribution of transmitted signals in light of the quality of service (QOS) associated with each signal.

Brief Summary Text (12):
The invention associates each signal packet with a particular transmission priority based on characteristics associated with each packet. The present invention schedules transmission of the cells of each packet based, at least in part, on the cell's transmission priority relative to transmission priorities associated with previously scheduled cells. Assigning a transmission priority to each virtual channel scheduled in the scheduling ring facilitates displacement of lower priority scheduling events and reorganization of the transmission schedule to ensure that the highest priority transmission events occur in a timely fashion.

Detailed Description Text (18):
In some cases, the ideal scheduling slot will be unoccupied, or occupied, but associated with a packet having a lower transmission priority than the packet 15 to be scheduled. In those cases, the closest-to-ideal scheduling slot is at the ideal scheduling slot. In other cases, however, the ideal scheduling slot may already be associated with a packet having the same or a higher transmission priority than the packet 15 to be scheduled. In that case, first cell 17a must be scheduled in the next closest slot after the ideal scheduling slot, which has a lower transmission priority than cell 17a. Once the closest-to-ideal scheduling slot has been identified, scheduling module 26 associates the virtual channel number corresponding to first cell 17a with the identified closest-to-ideal scheduling slot. This step may involve displacing virtual channel numbers corresponding to previously scheduled or rescheduled lower transmission priority cells and reorganization of scheduling ring 32.

Detailed Description Text (56):
The method begins at step 220, where scheduler 24 receives a scheduling request to schedule transmission of a cell of a previously unscheduled first packet. Details of the generation of this scheduling request will be explained in connection with FIG. 5b. Scheduling module 26 proceeds to schedule the transmission of the cell at step 240 based at least in part on inter-cell gap 82 and transmission priority 80 associated with the first packet. Details of the scheduling method will be explained in connection with FIGS. 5c and 5d.

Detailed Description Text (73):
Once the ideal scheduling slot is identified at steps 240b, scheduling module 26

proceeds to identify a closest-to-ideal scheduling slot by comparing at step 247 the transmission priority 80 associated with the previously unscheduled packet to the transmission priority associated with the identified ideal scheduling slot, which is stored in priority map 40. If scheduling module 26 determines at step 248 that the transmission priority associated with the packet is higher than the transmission priority associated with the ideal scheduling slot, scheduling module 26 sets the location of the closest-to-ideal scheduling slot to correspond with the location of the ideal scheduling slot at step 249. The transmission priority associated with the ideal scheduling slot may be lower than the transmission priority associated with the packet because, for example, the ideal scheduling slot is currently unoccupied, or because the packet previously associated with the ideal scheduling slot comprises a quality of service having a lower priority than packet sought to be scheduled.

Detailed Description Text (74):
If, on the other hand, scheduling module 26 determines at step 248 that the transmission priority associated with the previously unscheduled packet is equal to or lower than a transmission priority associated with the ideal scheduling slot, scheduling module 26 must locate the next closest slot after the ideal scheduling slot having a transmission priority lower than the transmission priority associated with the packet at step 250. From the previous description it can be appreciated that the closest-to-ideal scheduling slot may share the same location as the ideal scheduling slot, or may comprise a slot residing after the ideal scheduling slot, depending on the relative transmission priorities associated with the packet being scheduled and the ideal scheduling slot.

Detailed Description Text (75):
Scheduling module 26 locates the closest-to-ideal scheduling slot using priority map 40. To illustrate an exemplary method of traversing priority map 40 to locate the closest-to-ideal scheduling slot, a brief example will be given. Assume that scheduling ring pointer 34 currently points to slot 564 (which, in hexadecimal is 234, denoted 0x234) and that the ideal scheduling slot (determined with reference to next-time 76) is at slot 872 (0x368). Further assume that the transmission priority associated with the packet 15a desired to be scheduled is "medium" (priority 2). Scheduling module 26 wants to insert an entry for the virtual channel address associated with the cell being scheduled into scheduling ring 32 at the ideal scheduling slot (0x368), or as soon after that point as possible.

CLAIMS:

1. A method of scheduling transmission of a plurality of cells of a first signal packet associated with a first virtual channel address using a scheduling ring having a plurality of slots and a pointer operable to indicate a current slot, the method comprising: advancing the pointer to a slot associated with the first virtual channel address; initiating transmission of a previously scheduled first cell associated with the first virtual channel address; scheduling transmission of a previously unscheduled second cell associated with the first virtual channel address for transmission at a later time, wherein scheduling transmission of the previously unscheduled second cell comprises: identifying an ideal scheduling slot in the scheduling ring, a position of the ideal scheduling slot based at least in part on a desired value of an inter-cell gap and a transmission error associated with the first signal packet, the transmission error comprising a difference between a desired value of the inter-cell gap and a last value of the inter-cell gap, the last value of the inter-cell gap comprising a difference between the slot currently being serviced and the slot associated with a previously serviced cell of the first signal packet; identifying a closest-to-ideal scheduling slot in the scheduling ring comprising the closest slot at or after the ideal scheduling slot having a transmission priority lower than a transmission priority associated with the first signal packet; and associating the first virtual channel address with the

closest-to-ideal scheduling slot in the transmission schedule; and advancing the pointer to the next slot.

5. The method of claim 1, wherein identifying the closest-to-ideal scheduling slot comprises: identifying the closest-to-ideal scheduling slot as the ideal scheduling slot if the transmission priority of the first signal packet is higher than the transmission priority associated with the ideal scheduling slot; and identifying the closest-to-ideal scheduling slot as the next slot after the ideal scheduling slot having a lower transmission priority than the transmission priority of the first signal packet if the transmission priority of the first signal packet is equal or lower than the transmission priority associated with the ideal scheduling slot.

9. The method of claim 8, wherein scheduling transmission of the first cell of the second signal packet comprises: identifying an ideal scheduling slot in the scheduling ring based at least in part on an inter-cell gap associated with the second signal packet; and identifying a closest-to-ideal scheduling slot in the scheduling ring comprising the closest slot at or after the ideal scheduling slot having a transmission priority lower than a transmission priority associated with the second signal packet; and associating the first cell of the second signal packet with the closest-to-ideal scheduling slot.

12. The method of claim 9, wherein identifying the closest-to-ideal scheduling slot comprises: identifying the closest-to-ideal scheduling slot as the ideal scheduling slot if the transmission priority of the second signal packet is higher than the transmission priority associated with the ideal scheduling slot; and identifying the closest-to-ideal scheduling slot as the next slot after the ideal scheduling slot having a lower transmission priority than the transmission priority of the second signal packet if the transmission priority of the second signal packet is equal to or lower than the transmission priority associated with the ideal scheduling slot.

16. An integrated circuit operable to schedule transmission of a plurality of signal packets, comprising: a controller operable to receive control requests from a host memory and to generate scheduling requests based on the control requests received; a scheduling ring comprising a plurality of slots and a pointer operable to indicate a current slot; and a scheduler operable to advance the pointer to a slot associated with a first virtual channel address, to initiate transmission of a previously scheduled first cell of a signal packet associated with the first virtual channel address, and to schedule transmission of a previously unscheduled second cell of the signal packet for transmission at a later time; wherein the scheduler comprises a scheduling module operable to identify an ideal scheduling slot in the scheduling ring based at least in part on a desired inter-cell gap and a transmission error associated with the first signal packet, and to identify a closest-to-ideal scheduling slot in the scheduling ring comprising the closest slot at or after the ideal scheduling slot having a transmission priority lower than a transmission priority associated with the signal packet, the scheduling module further operable to associate the first virtual channel address with the closest-to-ideal scheduling slot; wherein the transmission error comprises a difference between a desired value of the inter-cell gap and a last value of the inter-cell gap, the last value of the inter-cell gap comprising a difference between the slot currently being serviced and the slot associated with a previously serviced cell of the first signal packet.

17. The integrated circuit of claim 16, wherein the scheduling ring comprises a memory containing the first virtual channel address and transmission characteristics including a desired inter-cell gap, a transmission priority, and a transmission error associated with the signal packet.

24. The integrated circuit of claim 16, further comprising a priority map

comprising a collection of transmission priorities associated with the slots of the scheduling ring, wherein the scheduler is operable to access the priority map to identify a closest-to-ideal slot based on a known position of an ideal slot and a transmission priority associated with the signal packet, the closest-to-ideal slot comprising the closest slot at or after the ideal slot having a transmission priority lower than the transmission priority associated with the signal packet.

26. A system for traffic shaping the transmission of a plurality of signal packets, comprising: a scheduling ring comprising a plurality of slots and a pointer operable to indicate a current slot; and a scheduler operable to advance the pointer to a slot associated with a first virtual channel address, to initiate transmission of a previously scheduled first cell of a signal packet associated with the first virtual channel address, and to schedule transmission of a previously unscheduled second cell of the signal packet for transmission at a later time; wherein the scheduler comprises a scheduling module operable to identify an ideal scheduling slot in the scheduling ring based at least in part on the desired inter-cell gap and a transmission error associated with the signal packet, and to identify a closest-to-ideal scheduling slot in the scheduling ring comprising the closest slot at or after the ideal scheduling slot having a transmission priority lower than a transmission priority associated with the signal packet, the scheduling module further operable to associate the first virtual channel address with the closest-to-ideal scheduling slot; wherein the transmission error comprises a difference between a desired value of the inter-cell gap and a last value of the inter-cell gap, the last value of the inter-cell gap comprising a difference between the slot currently being serviced and the slot associated with a previously serviced cell of the first signal packet.

27. The system of claim 26, wherein the scheduling ring comprises a memory containing the first virtual channel address and transmission characteristics including a desired inter-cell gap, a transmission priority, and a transmission error associated with the signal packet.

32. The system of claim 26, further comprising a priority map comprising a collection of transmission priorities associated with the slots of the scheduling ring, wherein the scheduler is operable to access the priority map to identify a closest-to-ideal slot based on a known position of an ideal slot and a transmission priority associated with the signal packet, the closest-to-ideal slot comprising the closest slot at or after the ideal slot having a transmission priority lower than the transmission priority associated with the signal packet.

36. The method of claim 34, wherein scheduling transmission of the previously unscheduled second cell comprises: identifying a closest-to-ideal scheduling slot in the scheduling ring comprising the closest slot at or after the ideal scheduling slot having a transmission priority lower than a transmission priority associated with the first signal packet; and associating the first virtual channel address with the closest-to-ideal scheduling slot in the transmission schedule.

38. The system of claim 37, wherein the scheduling module is operable to identify a closest-to-ideal scheduling slot in the scheduling ring comprising the closest slot at or after the ideal scheduling slot having a transmission priority lower than a transmission priority associated with the signal packet, the scheduling module further operable to associate the first virtual channel address with the closest-to-ideal scheduling slot.

42. The method of claim 40, wherein scheduling transmission of the second cell comprises: identifying a closest-to-ideal scheduling slot in the scheduling ring comprising the closest slot at or after the ideal scheduling slot having a transmission priority lower than a transmission priority associated with the first signal packet; and associating the first virtual channel address with the closest-to-ideal scheduling slot in the transmission schedule.

44. The system of claim 43, wherein the scheduling module is operable to identify a closest-to-ideal scheduling slot in the scheduling ring comprising the closest slot at or after the ideal scheduling slot having a <u>transmission priority lower than a transmission priority associated with the signal packet, the scheduling module further operable to associate</u> the first virtual channel address with the closest-to-ideal scheduling slot.

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L9: Entry 3 of 3 File: USPT Jul 1, 2003

DOCUMENT-IDENTIFIER: US 6587433 B1
TITLE: Remote access server for multiple service classes in IP networks

Brief Summary Text (9):
Naturally, any sort of service differentiation must be accompanied by an appropriate pricing scheme so that users will act efficiently with respect to the priority level they chose and thus the network resources that they consume. Network service providers are currently seeking more latitude in their service contracts with users. Implementing a differential service scheme according to pricing will give them this latitude.

Brief Summary Text (12):
In accordance with an illustrative embodiment of the invention, a method and device for implementing differential packet delivery services through a packet-based network is provided. According to the illustrative embodiment, a remote access server ("RAS") device provides differential packet delivery through the network according to a "per-hop behavior" or Differential Service Code Point (DSCP) field within the transmitted packets.

Detailed Description Text (21):
RAS 22 may also optionally include a number of management cards and router cards or other types of application cards necessary to implement system functions. Router card 212 can be employed as network egress cards. Data from connection sessions accessing the RAS 22 are sampled and converted to digital bitstreams. With the router card installed, the bitstreams are packetized and transmitted on the packet bus to the router card 210. The router card 210 removes any link-layer headers from each packet, such as Ethernet, HDLC, or PPP headers and transmits the resulting IP packet to an egress card. Alternatively, the modem cards 200 may contain knowledge of IP headers and transmit the IP packets directly to the RAS gateway. The network management card 212 can perform administrative tasks such as user authentication, accounting and logging.

Detailed Description Text (40):
In Table C above, the 6-bit Differential Service Code Point ("DSCP") or-Hop Behavior ("PHB") field specifies the general effect or priority that a router handling the packet should have on the packet. In an illustrative embodiment, the DSCP field is assigned 6-bits, but of course other numbers of bits may be used according to the desired application. Normal per-hop behavior is indicated with the value 000000, while expedited forwarding is indicated with the value 111000. Of course, with 6 available bits a wide range of priority or other behaviors (up to 64) can be provided. For example, Telnet can utilize a DSCP of 111000 while FTP operations may utilize 000000.

Detailed Description Text (59):
In another embodiment of the invention, stamping of the PHB is performed in the router card 22. When stamping is performed in the router card 210, the card must maintain an array of DS bytes for each modem port. When a bitstream is received from a modem card, the modem port can be determined either from framing information on the packet bus or from the IP source address of the IP packet when it is reassembled from the bitstream. Once an IP packet is isolated and buffered, the DS

byte can be stamped into the packet. The particular DS byte to stamp the packet can be determined by the current quality of service or precedence that the user has requested (and is paying for), the application(s) being used, the current billing rate or the flow information. The latter could be determined by TCP or UDP port number, current time of day, and source and destination IP addresses and port numbers, respectively, and transmitted to the RAS via control signaling (such as IPCP) on the dial-up link. All DS bytes must be given to the router card 210 by a user authentication and profiling device; i.e., either a RADIUS server 40 or a network management card 212. The router 210 maintains DS byte buffers for all active modems based on the modems' hardware address.

Generate Collection

1 of 2 File: USPT Dec 3, 2002

DOCUMENT-IDENTIFIER: US 6490705 B1
TITLE: Method and apparatus for receiving MPEG video over the internet

Abstract Text (1):
In order to transmit an inter-frame coded video signal, such as an MPEG-coded video signal, over a packet-based network such as the Internet, the video signal associated with at least one video frame, is split (102, 402) into a high priority partition and a low priority partition. A systematic forward error erasure/correction (FEC) code (108), such as a Reed Solomon (n,k) code, is then applied to bytes in the high priority partition. The forward error/erasure corrected high priority partition bytes and the low priority partition bytes are then combined (110) into n packets for transmission over the packet network to a receiver/decoder. Each of the n transmitted packets contains a combination of both high priority partition data bytes and low priority partition information bytes. In k of those packets the high priority partition data bytes are all high priority partition information bytes and in n-k of those packets all the high priority partition data byte are parity bytes produced by the FEC coding. More specifically, for each high priority partition byte position within the n packets, the forward error/erasure correction code is applied using one high priority partition information byte from the same byte position in each of those k packets to determine n-k parity bytes, which are arranged, one byte per packet, in the n-k packets containing high priority partition parity bytes. If up to n-k packets are lost in transmission over the packet network to the receiver (500, 600), then the high priority partition bytes in such lost packets can be recovered to applying FEC decoding (506) to the high partition bytes in the received packets. The most visually significant information is thus protected against packet loss over the network.

Application Filing Date (1):
19981022

Brief Summary Text (4):
With the exploding popularity of the public Internet in the past several years for transporting all types of data, there has been much recent interest in transmitting digitally encoded real-time audio and video over the Internet using the Universal Datagram Protocol (UDP). Because UDP is an unreliable protocol, network packet losses will likely occur and, as a result, will adversely affect the quality of the received audio and video. Recovery from packet losses may be performed solely by the receiver, or better quality can be achieved by involving both the sender and the receiver in the error recovery process. In networks that support prioritization, such as ATM, video quality can be improved in the presence of packet loss by using scalable video coding (see, e.g., R. Aravind, M. Civanlar, A. Reibman, "Packet Loss Resilience of MPEG-2 Scalable Video Coding Algorithms," IEEE Transactions on Circuits and Systems for Video Technology, Vol. 6, No. 5, October 1996). There is currently, however, no widespread support for prioritization on the public Internet. Overviews of proposed methods for error recovery for streaming of audio and video over the Internet, which involve both the sender and the receiver are disclosed by C. Perkins and O. Hodson in "Options for Repair of Streaming Media," Internet Engineering Task Force Intemet RFC 2354, June 1987,and G. Carle and E. Biersack in "Survey of Error Recovery Techniques for IP-Based Audio-Visual

Multicast Applications," IEEE Network, November/December 1997. While the general methods described in these overviews may be applicable to IP transmission of both audio and video, most of the studies published where specific techniques have been implemented involve audio only. Because of its higher data rates, and propagation of errors through inter-frame coding, it is more difficult to maintain video quality than audio, and audio techniques, therefore, cannot be directly applied to video signals.

Brief Summary Text (12):
In accordance with the present invention, an inter-frame coded video signal, such as an MPEG video signal, employs a data splitting function to split such a video stream into a high priority and a low priority partition. Systematic Forward Error/Erasure Correction coding is then performed on only the data in the high priority partition. The Forward Error/Erasure Corrected high priority partition data and the non-Forward Error/Erasure Corrected low priority partition data are then combined into packets and transmitted over the same network to a receiver, where they are decoded. Depending on the degree of protection against errors or erasures offered by the particular FEC code that is used, the loss of one or more packets containing high priority data can be corrected with no loss of data in the high priority partition. The effect of the loss of the low partition data in the lost packet or packets, which low partition is not protected, has much less of a deleterious effect on the quality of the decoded video signal than would the loss of data from the high priority partition data. Advantageously, by limiting the application of the Forward Error/Erasure Correction to only the higher priority partition data, and thus protecting against loss only that "more important data", the overhead requirement is reduced for protection against a given packet loss.

Brief Summary Text (13):
In the preferred embodiment, a Reed Solomon encoder is applied to the high priority data for an entire frame. For each RS(n,k) codeword, one information byte is taken from each of k packets and the constructed parity bytes are placed in h different packets, where n=k+h. Each individual frame's data is arranged in the n equal length packets that contain a combination of: packet headers; high priority data comprising one of either information bytes or parity bytes; and low priority data bytes, the latter comprising only information bytes since no error-correction coding is performed on the low priority data. The same number of bytes of high priority data (information or parity in any one packet) are placed in each of the n equal length packets, and the same number of bytes of low priority data (information only) are placed in these same n packets, which together represent the video frame. Amongst these n equal length packets, k packets only contain high priority partition information bytes and h packets only contain the high priority parity bytes. The parity byte in each high priority byte position in each of theses h packets is formed from the RS(n,k) code as it is applied to the k high priority partition information bytes in a corresponding byte position in the k other high priority partition information-containing packets associated with the frame. Advantageously, arranging the packets in this manner minimizes the amount of overhead and delay for a given packet loss protection rate.

Brief Summary Text (14):
A receiving decoder, upon receiving the packets associated with a frame separates the high priority partition bytes and low priority partition bytes in each packet according to the numbers of such bytes or each type included within each packets, which numbers are transmitted in the packet headers. RS(n,k) decoding is applied byte position-by-byte position across the high priority partition portion within the received packets. If up to h of the n frame packets are lost, the RS decoding process recovers each high priority byte in the lost packet or packets. Full reconstruction of the high priority partition information bytes that were transmitted in the k packets of the n packets that contained high priority partition data is thus effected. Although the low priority partition data in the lost packets is unrecoverable, the fully recovered high priority partition data

enables the video picture to be decoded, albeit in what might be at a reduced quality level for that frame or that portion of a frame in which only the high priority partition information is available.

Detailed Description Text (2):
The MPEG-2 standard (ITU-TRecommendation H.262, "GENERIC CODING OF MOVING PICTURES AND ASSOCIATED AUDIO INFORMATION", July 1995) includes several means for performing scalable video coding, including spatial scalability, SNR scalability, and data partitioning. In scalable video coding, receivers with different available bandwidths can receive and decode appropriate representations of coded video, by receiving a base layer, and if bandwidth is available, receiving one or more enhancement layers. The more layers received, the higher the quality of the decoded video. In the aforenoted paper by Aravind, Civanlar and Reibman, data partitioning was applied to provide protection against packet loss for transmitting an MPEG-coded video signal in a network that supports prioritization such as ATM. Specifically, protection against packet loss was shown to be achievable by transmitting the base layer at a higher priority than the enhancement layer(s). Since the public Internet does not currently support prioritization of packets this technique cannot be applied to the transmission of coded video over the Internet.

Detailed Description Text (3):
The inventor has recognized, however, that an advantage can be achieved by using, in a non-prioritized network, such as the public Internet, a data splitting functionality that is used in the prior art for packet protection over a network that does support prioritization. By incorporating a data splitting functionality together with a Forward Error/Erasure Correction functionality for transmitting an inter-frame compression-coded video signal over a non-prioritized public Internet, the amount of overhead needed for packet protection is reduced, while achieving the improvement in the subjective video quality that packet protection affords. Further, by combining high priority and low priority data in the same packets, the delay for equal overhead and protection is advantageously reduced.

Detailed Description Text (4):
FIG. 1 is a first embodiment of an encoder in accordance with the present invention. In this embodiment of the present invention, an encoder which is compliant with MPEG-standardized data partitioning is used to split an incoming video data stream into a high priority partition and a low priority partition. With reference to FIG. 1, an incoming video data stream on 101 is inputted to a such a compliant data partitioning MPEG-standardized encoder 102 which, in accordance with the standard, such as the MPEG-2 data partitioning standard, compression-codes the input video bitstream and splits the compression-coded bitstream into two output layers. The first layer is the base layer, referred to herein as the high priority (HP) partition. The second layer is the enhancement layer, referred to herein as the low priority (LP) partition.

Detailed Description Text (5):
As is well known to one skilled in the art of MPEG coding, an MPEG-coded video bitstream includes headers at the sequence level, at the group-of-picture (GOP) level, at the picture (frame) level, and at the slice level. As is well known, a slice consists of a group of adjacent macroblocks, where each macroblock in itself consists of a group of four adjacent luminance blocks of data and two chrominance blocks. At the picture level, a frame is classified as being as an intra-frame coded (I) frame, an inter-frame coded predictive (P) frame, or a bi-directional inter-frame predictive (B) frame. At the macroblock level, for the predictive-type P and B type frames, information is included which indicates whether the macroblock is inter or intra coded with respect to another frame, as well as motion vector information. The information at the block level includes low frequency and high frequency discrete cosine transformation (DCT) coefficients derived from actual pixel information.

Detailed Description Text (6):
In accordance with MPEG standards, data partitioning can be effected at a plurality of different priority breakpoints, such as above the macroblock layer, at the macroblock layer, or within the macroblock layer. Generally, the more critical parts of the coded bitstream, such as the headers, the motion vectors and the low frequency DCT coefficients, are allocated to the HP partition, and the less critical data, such as the higher frequency DCT coefficients, are allocated to the LP partition. In accordance with the standard, a priority breakpoint can be chosen for each slice, which determines which codeword types are arranged into which partition.

Detailed Description Text (7):
In the embodiment of the invention in FIG. 1, which uses the standardized data-partitioning MPEG encoder 102, priority breakpoints are determined in accordance with the type of frame (I, P, or B) in the data stream that is being partitioned. Specifically, in this embodiment, for each I frame, all the data is placed in the HP partition, with no data being placed in the LP partition. Other embodiments may divide the data from each I frame between the HP and LP partitions. For each B frame, as much frame data as the MPEG standards relating to data partitioning allows is placed in the LP partition, with the remaining data placed in the HP partition. For each P frame, frame data is divided between the HP and LP partitions so that data elements through the first two DCT coefficients of each block are placed in the HP partition, with the higher order DCT coefficients placed in the LP partition. Different priority breakpoints in the three frame types between the LP and HP partitions other than described above for this embodiment could equally be used. It would be expected, however, that a higher breakpoint would be used for I frames than for P frames, which, in turn, would be higher than the breakpoint for B frames. In this standards-compliant embodiment, sequence, GOP, and picture headers are copied into both partitions for error resilience.

Detailed Description Text (10):
FIG. 2 shows an example of an arrangement of a frame group of packets for an RS (4, 3) code. As can be noted, three packets (k=3) contain HP information bytes information bytes and LP information bytes. One packet (h=1) contains the HP parity bytes and the LP information bytes. Arranging the packets in the manner illustrated by this figure minimizes the amount of overhead for a given packet loss protection rate without adding to delay. The packet header in each packet contains the packet number, the number of the packets in the frame group of packets (n), the number of packets with parity data in the frame group of packets (h), the temporal reference value for the frame, the frame type (I, P, or B), the HP/LP priority breakpoints, and the number of HP and LP bytes in each packet.

Detailed Description Text (14):
For the example of FIG. 2, if none of the packets containing the frame information are lost in transmission, the decoded video will be perfectly decoded (i.e., identical to that encoded). If one of the packets (or up to h packets in the general case for R(n,k) coding) is lost, then all of the HP information in that packet is recoverable using a Reed Solomon decoder. The LP information data in the lost packet could not, however, be recovered since it is not protected. The portions of the picture that correspond to the macroblocks whose LP data was received will, however, be perfectly decoded, and those that correspond to macroblocks whose LP data is lost will decode only the HP data for those macroblocks. Those macroblocks which are decoded using only HP data may be noticeable to a viewer, but are unlikely to be visually offensive as long as the HP/LP priority breakpoint is properly chosen. The exact visual quality of macroblocks decoded using only HP data depends on this HP/LP priority breakpoint, as well as the characteristics of the video source material. In the embodiment of FIG. 1 in which a standardized data partitioning MPEG encoder 102 is used to compression-code the input video stream and form the HP and LP partitions, the lowest level header copied into both partitions is the picture header. Thus, once

any packet is lost, and with it the LP partition information contained therein, which cannot be recovered, the LP partition data that is received in the next received packets cannot be properly incorporated into the decoding process since there is no identifiable spatial entry point with which that received data can be associated until a next picture header is received. Thus, with the embodiment of FIG. 1, the perceptual effect of lost LP partition information will extend until a packet is received that contains a next picture header within the received LP partition information. Once that picture header is received, the LP partition data that follows can be properly incorporated with the received HP partition data.

Detailed Description Text (20):
As previously described, the data partitioning MPEG encoder 102 partitions I frames so that all frame data is placed in the HP partition to minimize the effect of an error in an I frame from propagating to other frames. Since B frames are not used for prediction, as much data as is possible via the data partitioning standard is placed in the LP partition. In the embodiment of FIG. 4, all data for all B frames is placed in the LP partition as opposed to the embodiment of FIG. 1 in which some data, in accordance with the standards, is required in the HP partition. In both the embodiment of FIG. 1 and FIG. 4, the P frame data is divided between the HP and LP partitions in the manner described above. Within each P frame macroblocks can be either inter-frame or intra-frame coded. In the embodiment of FIG. 4, different priority breakpoints are chosen for inter and intra coded macroblocks in each P frame, which the data partitioning MPEG encoder 102 in FIG. 1 could not do. Inter-coded macroblocks decoded using only the HP partition (and not the LP partition) may retain high frequency information from the corresponding motion-compensated macroblocks in the previous frame, while intra-coded macroblocks may not. Hence, it is desirable to set the priority breakpoint for intra-coded macroblocks to include more DCT coefficients in the HP partition than are included in inter-coded macroblocks. Advantageously, this reduces the overhead rate for given level of quality or improves the quality for the same overhead rate.

Detailed Description Text (24):
A decoder network 500 that is associated with the encoder network of FIG. 1 is illustrated in FIG. 5. As a data partitioning MPEG encoder 102 is incorporated within the encoder network of FIG. 1, the decoder network of FIG. 5 incorporates a complimentary data partitioning MPEG decoder 510. In FIG. 5, the serial packets transmitted by the encoder network over the UDP/IP network 501 are inputted to a de-packetizer/decoder 502. De-packetizer/decoder 502 includes a de-packetizer 503 which receives the serial packets and strips the header information in each packet and passes it to a processor 504. The packet header information includes: a packet number; a frame number; the type of frame (I, B, P); the (n,k) parameters which define the packet structure of the frame; and the number of HP partition bytes, HPB, and LP partition bytes, LPB, in each packet. Processor 504, from the header information, determines the start of the frame, and from the parameter n, "knows" that that many packets are used to define the frame. Further, from the packet numbers received, processor 504 determines which particular of those n packets, if any, are missing and their position within the sequence of these n packets. In response to receiving all such information from processor 504, de-packetizer 503 strips off the packet header of each packet within the frame and divides the data in each packet into its HP and LP portions. For those packets which are determined to be missing, de-packetizer 503 inserts "0" bytes or an error code in the HP and LP data streams. The HP serial byte stream output of de-packetizer 503 consists of n sub-packets-worth of HP data, each sub-packet containing HPB bytes. This HP stream is inputted to an interleaver 505 to decode the RS(n,k) encoded words that exist across the sub-packets and to replace any missing data from up to h lost packets. Thus, for each byte position across the sub-packet, a byte is selected from each such sub-packet at that byte position to form an input word to an RS decoder 506.

Detailed Description Text (29):

Data partitioning MPEG decoder 510, in response to the inputted LP partition data and the HP partition data decompresses and reformulates the transmitted video data in accordance with its standardized algorithm. Where, with respect to specific pels in the frame, corresponding HP data is available (by being actually received or recovered) but LP data is not available, the subjective video quality of the reconstructed video frame is degraded. Furthermore, that spatial portion of the reconstructed video frame that follows, in a scanning sense, those pels in the frame associated with the lost LP partition data, is also degraded since the LP partition data in the next received packet cannot be associated with specific spatial points within the frame and thus with the HP data. As noted, only picture headers are included within both the HP and LP partitions. Thus, until the next picture header is received, all LP partition data within the video frame that follows a lost packet will be unable to be combined with spatially corresponding HP data to decode and reconstruct the video signal. Further, since the type of frames in this embodiment which are divided into separate HP and LP partitions are the P frames, which are used to predict the next frame, the loss of LP data for reconstructing the remainder of the frame will have an effect on the quality of the next P and B frames, albeit at a substantially reduced level as compared to the effect that a total loss of data would have caused without the present invention in which the more important HP partition data has been protected. In the event that more than h packets are lost in transmission, both HP and LP partition data is lost and not recoverable. Standard error concealment techniques can then be used to minimize the degradation in video quality.

Other Reference Publication (2):
Anand et al., "FEC and Priority for VBR Video Distribution over ATM", 1993 Canadian Conf. on Electrical and Computer Engineering, pp. 550-553.*

Other Reference Publication (7):
M. Andronico et al., "Performance Analysis of Priority Encoding Transmission of MPEG Video Streams," IEEE Globecom 1996, Nov. 18-22,1996, vol. 1, 267-271.

Other Reference Publication (15):
A. Albanese, J. Blomer, J. Edmonds, M. Luby, and M. Sudan, "Priority Encoding Transmission", IEEE Transactions on Information Theory, vol. 42, No. 6, Nov. 1996, pp. 1737-1744.

CLAIMS:

1. A method of decoding a compression-coded video signal that has been packetized and transmitted over a packet-based network in a sequence of packets that are associated with at least one frame of the compression-coded video signal, the method comprising the steps of: receiving packets associated with the at least one frame of the compression-coded video signal, each of the received packets containing both low priority partition information bytes and forward error/erasure correction (FEC)-coded high priority partition data bytes associated with the at least one frame of the compression-coded video signal, the compression-coded video signal having been split into low priority and high priority partitions before being packetized and transmitted; determining which transmitted packets associated with the at least one frame of the video sequence have not been received; applying a FEC decoding to the FEC-coded high priority partition data bytes in the received packets to determine high priority partition information bytes associated with the at least one frame, the step of applying a FEC decoding reconstructing high priority partition information bytes in those packets determined not to have been received; combining the received low priority partition information bytes and the decoded and reconstructed high priority partition information bytes; and decompression-decoding the combined received low priority partition information bytes and the decoded and reconstructed high priority partition information bytes to reproduce the at least one frame of the video signal.

3. The method of claim 1 wherein the received FEC-coded high priority partition data bytes are coded with a systematic forward error/erasure correction code, the received FEC-coded high priority partition data bytes comprising a combination of the high priority partition information bytes associated with the at least one frame and associated parity bytes.

4. The method of claim 3 wherein each received packet has an equal length, the high priority partition data bytes in each received packet comprising either all high priority partition information bytes associated with the at least one frame or all parity bytes.

5. The method of claim 4 wherein an equal number of low priority partition information bytes are in each received packet and an equal number of high priority partition data bytes are in each received packet.

6. The method of claim 5 wherein for each high priority byte position associated with the received packets, high priority partition information bytes in the same high priority byte position from each received packet containing high priority partition information bytes, one byte per packet, are associated with at least one parity byte in the same byte position in one or more received packets containing parity bytes, the FEC decoding being applied to the high priority partition information bytes and associated at least one parity byte at each high priority byte position, one byte position at a time, to determine high priority partition information bytes in each high priority byte position in a packet that was not received.

8. The method of claim 6 further comprising the steps of: determining the number of packets (n) in which the at least one frame of the compression-coded video signal has been packetized and transmitted; and determining of the n transmitted packets, the number of packets (k) transmitted containing both high priority partition information bytes associated with the at least one frame and low priority partition information bytes.

10. The method of claim 8 wherein the received packets associated with the at least one video frame of the video signal are protected against a packet loss over the packet-based network of high priority partition information bytes in up to (n-k) packets.

12. A decoder for decoding a compression-coded video signal that has been packetized and transmitted over a packet-based network in a sequence of packets that are associated with at least one frame of the compression-coded video signal, the decoder comprising: a depacketizer connected to receive packets associated with the at least one frame of the compression-coded video signal, each of the received packets containing both low priority partition information bytes and forward error/erasure correction (FEC)-coded high priority partition data bytes associated with the at least one frame of the compression-coded video signal, the compression-coded video signal having been split into low priority and high priority partitions before having been packetized and transmitted, the depacketizer separating the FEC-coded high priority partition data bytes and the low priority partition information bytes; a FEC decoder connected to receive and decode the FEC-coded high priority partition data bytes; and a video decompression decoder connected to receive the depacketized low priority partition information bytes and the FEC-decoded high priority partition data bytes, the video decompression decoder combining and decompression-decoding the low priority partition information bytes and the FEC-decoded high priority partition data bytes.

14. The decoder of claim 12 wherein the FEC-coded high priority partition data bytes are coded with a systematic FEC code, the received FEC-coded high priority partition data bytes comprising a combination of the high priority partition information bytes associated with the at least one frame and associated parity

bytes.

15. The decoder of claim 14 wherein each received packet has an equal length, the high priority partition data bytes in each received packet comprising either all high priority partition information bytes associated with the at least one frame or all parity bytes.

16. The decoder of claim 15 wherein an equal number of low priority partition information bytes are in each received packet and an equal number of high priority data bytes are in each received packet.

17. The decoder of claim 16 wherein for each high priority byte position associated with the received packets, high priority partition information bytes in the same high priority byte position from each received packet containing high priority partition information bytes, one byte per packet, are associated with at least one parity byte in the same byte position in one or more received packets containing parity bytes, the FEC decoder decoding the high priority partition information bytes and associated at least one parity byte at each high priority byte position, one byte position at a time, to determine high priority partition information bytes in each high priority byte position in a packet that was not received.

19. The decoder of claim 17 further comprising a processor, the processor: determining the number of packets (n) in which the at least one frame of the compression-coded video signal has been packetized and transmitted; and determining of the n transmitted packets, the number of packets (k) transmitted containing both high priority partition information bytes associated with the at least one frame and low priority partition information bytes.

21. The decoder of claim 19 wherein the received packets associated with the at least one video frame of the video signal are protected against a packet loss over the packet-based network of high priority partition information bytes in up to (n-k) packets.

23. A decoder for decoding a compression-coded video signal that has been packetized and transmitted over a packet-based network in a sequence of packets that are associated with at least one frame of the compression-coded video signal, the decoder comprising: means for receiving packets associated with the at least one frame of the compression-coded video signal, each of the received packets containing both low priority partition information bytes and forward error/erasure correction (FEC)-coded high priority partition data bytes associated with the at least one frame of the compression-coded video signal, the compression-coded video signal having been split into low priority and high priority partitions before being packetized and transmitted; means for determining which transmitted packets associated with the at least one frame of the video sequence have not been received; means for applying a FEC decoding to the FEC-coded high priority partition data bytes in the received packets to determine high priority partition information bytes associated with the at least one frame, the means for applying a FEC decoding reconstructing high priority partition information bytes in those packets determined not to have been received; means for combining the received low priority partition information bytes and the decoded and reconstructed high priority partition information bytes; and means for decompression-decoding the combined received low priority partition information bytes and the decoded and reconstructed high priority partition information bytes to reproduce the at least one frame of the video signal.

25. The decoder of claim 23 wherein the received FEC-coded high priority partition data bytes are coded with a systematic forward error/erasure correction code, the received FEC-coded high priority partition data bytes comprising a combination of the high priority partition information bytes associated with the at least one frame and associated parity bytes.

26. The decoder of claim 25 wherein each received packet has an equal length, the high priority partition data bytes in each received packet comprising either all high priority partition information bytes associated with the at least one frame or all parity bytes.

27. The decoder of claim 26 wherein an equal number of low priority partition information bytes are in each received packet and an equal number of high priority partition data bytes are in each received packet.

28. The decoder of claim 27 wherein for each high priority byte position associated with the received packets, high priority partition information bytes in the same high priority byte position from each received packet containing high priority partition information bytes, one byte per packet, are associated with at least one parity byte in the same byte position in one or more received packets containing parity bytes, the FEC decoding being applied to the high priority partition information bytes and associated at least one parity byte at each high priority byte position, one byte position at a time, to determine high priority partition information bytes in each high priority byte position in a packet that was not received.

30. The decoder of claim 28 further comprising processing means for: determining the number of packets (n) in which the at least one frame of the compression-coded video signal has been packetized and transmitted; and determining of the n transmitted packets, the number of packets (k) transmitted containing both high priority partition information bytes associated with the at least one frame and low priority partition information bytes.

32. The decoder of claim 30 wherein the received packets associated with the at least one video frame of the video signal are protected against a packet loss over the packet-based network of high priority partition information bytes in up to (n-k) packets.

34. A method of decoding a compression-coded video signal that has been packetized and transmitted over a packet-based network in a sequence of packets that are associated with at least one frame of the compression-coded video signal, the method comprising the steps of: receiving packets associated with the at least one frame of the compression-coded video signal, each of the received packets containing both forward error/erasure correction (FEC)-coded high priority partition data bytes that have been coded with a systematic FEC code, and low priority partition information bytes associated with the at least one frame of the compression-coded video signal, the compression-coded video signal having been split into high priority and low priority partitions before having been packetized and transmitted; determining which transmitted packets associated with the at least one frame of the video sequence have not been received; applying a FEC decoding to the FEC-coded high priority partition data bytes in the received packets to determine high priority partition information bytes associated with the at least one frame, wherein for each high priority byte position associated with the received packets, high priority partition information bytes in the same high priority byte position from each received packet containing high priority partition information bytes, one byte per packet, are associated with at least one parity byte in the same byte position in one or more received packets containing parity bytes, the step of applying a FEC decoding further comprising applying a FEC decoding to the high priority partition information bytes and associated at least one parity byte at each high priority byte position, one byte position at a time, to reconstruct high priority partition information bytes in each high priority byte position in a packet that was determined not have been received; combining the received low priority partition information bytes and the decoded and reconstructed high: priority partition information bytes; and decompression-decoding the combined received low priority partition information bytes and the decoded and reconstructed

high priority partition information bytes to reform the at least one frame of the video signal.

35. The method of claim 34 wherein each packet has an equal length and an equal number of low priority partition information bytes are in each packet and an equal number of high priority partition information bytes or parity bytes are in each packet.

37. The method of claim 35 further comprising the steps of: determining the number of packets (n) in which the at least one frame of the compression-coded video signal has been packetized and transmitted; and determining of the n transmitted packets, the number of packets (k) transmitted containing both high priority partition information bytes associated with the at least one frame and low priority partition information bytes.

39. The method of claim 37 wherein the received packets associated with the at least one video frame of the video signal are protected against a packet loss over the packet-based network of high priority partition information bytes in up to (n-k) packets.

41. A decoder for decoding a video signal for transmission over a packet-based network comprising: means for receiving packets associated with the at least one frame of the compression-coded video signal, each of the received packets containing both forward error/erasure correction (FEC)-coded high priority partition data bytes that have been coded with a systematic FEC code, and low priority partition information bytes associated with the at least one frame of the compression-coded video signal, the compression-coded video signal having been split into high priority and low priority partitions before having been packetized and transmitted; means for determining which transmitted packets associated with the at least one frame of the video sequence have not been received; means for applying a FEC decoding to the FEC-coded high priority partition data bytes in the received packets to determine high priority partition information bytes associated with the at least one frame, wherein for each high priority byte position associated with the received packets, high priority partition information bytes in the same high priority byte position from each received packet containing high priority partition information bytes, one byte per packet, are associated with at least one parity byte in the same byte position in one or more received packets containing parity bytes, the means for applying a FEC decoding applying a FEC decoding to the high priority partition information bytes and associated at least one parity byte at each high priority byte position, one byte position at a time, to reconstruct high priority partition information bytes in each high priority byte position in a packet that was determined not have been received; means for combining the received low priority partition information bytes and the decoded and reconstructed high priority partition information bytes; and means for decompression-decoding the combined received low priority partition information bytes and the decoded and reconstructed high priority partition information bytes to reform the at least one frame of the video signal.

42. The decoder of claim 41 wherein each packet has an equal length and an equal number of low priority partition information bytes are in each packet and an equal number of high priority partition information bytes or parity bytes are in each packet.

44. The decoder of claim 42 further comprising a processor, the processor: determining the number of packets (n) in which the at least one frame of the compression-coded video signal has been packetized and transmitted; and determining of the n transmitted packets, the number of packets (k) transmitted containing both high priority partition information bytes associated with the at least one frame and low priority partition information bytes.

46. The decoder of claim 44 wherein the received packets associated with the at least one video frame of the video signal are protected against a packet loss over the packet-based network of high priority partition information bytes in up to (n-k) packets.

48. The method of claim 6 further comprising the step of providing to the step of decompression-decoding the low priority partition information bytes and the high priority partition information bytes, information indicating which low priority partition information bytes are missing as a result of having been transmitted in a packet determined not to have been received.

49. The decoder of claim 17 wherein the video decompression decoder is provided with information indicating which low priority partition information bytes are missing as a result of being transmitted in a packet not received.

50. The decoder of claim 28 wherein the means for decompression-decoding the low priority partition information bytes and the high priority partition information bytes is provided with information indicating which low priority information bytes are missing as a result of having been transmitted in a packet determined not to have been received.

51. The method of claim 34 further comprising the step of providing to the step of decompression-decoding the low priority partition information bytes and the high priority partition information bytes, information indicating which low priority partition information bytes are missing as a result of having been transmitted in a packet determined not to have been received.

52. The decoder of claim 41 wherein the means for decompression-decoding the low priority partition information bytes and the high priority partition information bytes is provided with information indicating which low priority information bytes are missing as a result of having been transmitted in a packet determined not to have been received.

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L11: Entry 30 of 41 File: USPT Jan 11, 2000

DOCUMENT-IDENTIFIER: US 6014694 A
TITLE: System for adaptive video/audio transport over a network

Abstract Text (1):
A system for adaptively transporting video over networks wherein the available bandwidth varies with time. The system comprises a video/audio codec that functions to compress, code, decode and decompress video streams that are transmitted over networks having available bandwidths that vary with time and location. Depending on the channel bandwidth, the system adjusts the compression ratio to accommodate a plurality of bandwidths ranging from 20 Kbps for POTS to several Mbps for switched LAN and ATM environments. Bandwidth adjustability is provided by offering a trade off between video resolution, frame rate and individual frame quality. The system generates a video data stream comprised of Key, P and B frames from a raw source of video. Each frame type is further comprised of multiple levels of data representing varying degrees of quality. In addition, several video server platforms can be utilized in tandem to transmit video/audio information with each video server platform transmitting information for a single compression/resolution level.

Application Filing Date (1):
19970626

Brief Summary Text (11):
The system functions to generate a prioritized video data stream comprising multiple levels from a raw source of video. This video stream is stored in a file and accessed by the video server when servicing clients. In operation, the video client only receives a subset of the levels. The levels are chosen to have a suitable data content to match that of the network connection. This permits a better fit between network bandwidth consumed and video image quality. Each of the levels is built on top of the previous levels, with the higher levels providing incremental information not present in the lower levels. This ensures that bandwidth is not wasted on the client end or on the encoder/server side. The system generates the video stream that is sent to the client such that a loss of any individual packet on the network will not cause sustained deged quality at the client

Brief Summary Text (12):
The scaleable compression performed by the system is suitable for transparent video within an Internet environment characterized by large diversity and heterogeneity. The system functions to match the image quality of the video data being transported with the wide variations in available network bandwidth. In addition, the system can adjust the video data to match the differences in available computing power on the client computer system. The system, utilizing `best effort` protocols such as those found on the Internet, adapts to the time varying nature of the available bandwidth.

Brief Summary Text (14):
The step of compressing comprises the step of compressing the raw video source into a plurality of different types of frames, each frame type containing different amount of video content information, the plurality of different types of frames grouped so as to form a video stream consisting of a plurality of group of pictures

(GOP) sequences. The step of compressing comprises the step of compressing the raw video source into Key, ~~P and B type frames,~~ the Key, P and B frames generated so as to form a video steam ~~consisting of a plurality~~ of group of pictures (GOP) sequences.

Brief Summary Text (15):
There is also provided in accordance with the present invention a method of transporting video from a video server to a video client over a network channel, comprising the steps of compressing data from a raw video source so as to generate a plurality of frames, each frame being of a particular frame type, each frame type containing a particular amount of video content information, each frame comprising ~~a plurality of levels, each level corresponding to a particular~~ degree of compression, estimating the bandwidth of the network channel, deter the amount of video information waiting to be displayed at the video client, selecting one of the plurality of levels of each frame to send over the network channel in accordance with the bandwidth estimate whereby the level selected optimizes the use of the bandwidth of the network channel, choosn which frames having a particular frame type to send over the network channel in accordance with the amount of video information waiting to be displayed at the video client, and sending the chosen frames having a particular frame type and of the selected level over the network channel.

Brief Summary Text (16):
Further, there is provided in accordance with the present invention a video server for transporting video from a video source over a network channel to a video client, the video source consisting of a plurality of frames of video data, each frame of video data consisting of multiple compression levels and being of a particular type, the video server comprising receiver means for inputting frames of video data from the video source, sending means coupled to the receiver means, the sending means for determining which compression level within the frame and which frames having a particular type to transmit in accordance with the estimated available bandwidth of the network channel, the sending means for encapsulating the frames of video data into a plurality of packets for transmission over the network channel, and a controller for managing the operation of the receiver means and the sending means whereby the rate of transmission of the sending means is maintained so as to match the available bandwidth of the network channel.

Brief Summary Text (17):
In addition, the sending means comprises a rate control unit for measuring the available bandwidth of the network channel, a frame selector for inputting video frame data output by the receiver means, the frame selector outputting frames of a particular compression level in accordance with the bandwidth measured by the rate control unit, a packet generator for inputting video frame data output by the frame selector, the packet generator for encapsulating the video frame data into a plurality of packets for transmission, the packet generator determining which frames having a particular type are to be transmitted, a packet transmitter for placing onto the network channel the plurality of packets output by the packet generator, and a receiver for receiving acknowledgments sent by the video client over the network channel in response to packets received thereby.

Brief Summary Text (20):
There is also provided in accordance with the present invention a method of transporting video from a video server to a video client over a network channel, comprising the steps of compressing data from a raw video source so as to generate a plurality of frames, each frame being of a particular frame type, each frame type containing a particular amount of video content information, each frame comprising a plurality of levels, each level corresponding to a particular degree of compression, estimating the bandwidth of the network channel, determining the amount of video information waiting to be displayed at the video client, selecting one of the plurality of levels of each frame to send over the network channel in

accordance with the bandwidth estimate whereby the level selected optimizes the use of the bandwidth of the network channel, choosing which frames having a particular frame type to send over the network channel in accordance with the amount of video information waiting to be displayed at the video client, sending the chosen frames of a type containing a higher amount of video data content and of a selected level over the network channel utilizing a reliable communication protocol, and sending the chosen frames of a type containing a lower amount video data content and of a selected level over the network channel utilizing an unreliable communication protocol.

Brief Summary Text (21):
Still farther, there is provided in accordance with the present invention a video server system for transporting video from a plurality of video sources over a network channel to a video client, each video source consisting of a plurality of frames of video data, each frame of video data consisting of a single compression level and being of a particular type, the video server system comprising a plurality of video servers, each video server associated with a single video source at a particular compression level, each video server comprising receiver means for inputting fames of video data from the video source associated with that particular video server, sending means coupled to the receiver means, the sending means for determining which frames having a particular type to transmit in accordance with the available bandwidth of the network channel, the sending means for encapsulating the frames of video data into a plurality of packets for transmission over the network channel, a controller for managing the operation of the receiver means and the sending means, and a rate controller for determining which video server to utilize for transmission of video data based on the available bandwidth of the network channel.

Brief Summary Text (22):
The sending means comprises means for interfacing the video server to the rate controller, a bandwidth measurement unit for measuring the available bandwidth of the network channel, a packet generator for inputting video frame data output by the receiver means, the packet generator for encapsulating the video frame data into a plurality of packets for transmission, the packet generator determining which frames having a particular type are to be transmitted, a packet transmitter for placing onto the network channel the plurality of packets output by the packet generator, and a receiver for receiving acknowledgments sent by the video client over the network channel in response to packets received thereby.

Detailed Description Text (7):
The video compression/file generator 14 in combination with the video client 22 comprise a video/audio codec or coder/decoder that functions to compress, code, decode and decompress video streams that are transmitted over the network 20 into a compressed video and audio file. The compressed file may be in any suitable format such as Audio Video Interleaved (AVI) format. Note that the network may comprise any type of network, TCP/IP or otherwise including the Internet. The generation of the compressed video and audio file 16 can be performed either online or off-line. Typically, the video and audio file is generated off-line. Note that, any suitable method of video compression can be utilized in the present invention such as described in connection with the Motion Pictures Expert Group (MPEG)-1, MPEG-2 or MPEG-4 standards.

Detailed Description Text (9):
The system functions to generate a prioritized video data stream comprising multiple levels from a raw source of video 12. This video steam is stored in a file (compressed video and audio file 16 in FIG. 1) and accessed by the video server IS when servicing clients 22. In operation, the video client only receives a subset of the levels that form the video and audio file 16. The levels are chosen to have a suitable data content to match that of the network connection between server and client. This permits a better fit between network bandwidth consuned and video

image quality. Each of the levels is built on top of the previous levels, with the higher levels providing incremental information not present in the lower levels. This ensures that bandwidth is not wasted on the client end or on the encoder/server side. The system generates the video stream that is sent to the client such that a loss of any individual packet on the network will not cause sustained degraded quality at the client.

Detailed Description Text (10):
The scaleable compression performed by the system is suitable for transparent video within an Internet environment characterized by large diversity and heterogeneity. The system functions to match the image quality of the video data being transported with the wide variations in available network bandwidth. In addition, the system can adjust the video data to match the differences in available computing power on the client computer system. The system, utilizing `best effort` protocols such as those found on TCP/IP networks, adapts to the time varying nature of the available bandwidth.

Detailed Description Text (18):
The generation of the video source file, e.g., video and audio file 16 (FIG. 1), and its internal format will now be described in more detail. As previously described, the video source file used by the video server to generate the video stream that is sent over the network connection to the client is created by the video compression/file generator 14 (FIG. 1). The input to the compression/generator is a raw video source 12. The raw video source can be, for example, a non compressed AVI file, a non compressed QuickTime file or a compressed MPEG-1 audio/video file.

Detailed Description Text (19):
The function of the video compression/file generator is to compress the raw video source into multiple levels of varying quality. In particular, the raw video source is compressed into three types of data objects commonly referred to as frames. The three types of frames include Key frames, P frames and B frames. These frames are similar to the I frames, P frames and B frames, respectively, as described in the MPEG-1 specification standard (officially designated as ISO/IEC 11172) and the MPEG-2 specification standard (officially designated as ISO/IEC 13818).

Detailed Description Text (22):
With reference to FIG. 4, the example GOP is shown comprising a Key frame 60, three B fames 62, 66, 70 and three P frames 64, 68, 72. Each GOP typically represents a particular unit or chunk of video information such as a scene in the video. For example, depending on the compression technique used, drastic scene changes may trigger the generation of a new GOP headed by a new Key frame. The video stream, as shown by the arrow, is made up of a sequence of GOPs transmitted one after the other. Each of the three types of frames will now be described in more detail.

Detailed Description Text (24):
The video data incorporated into P frames includes data that is predicted based on a previous Key frame or a previous P frame. The information that is included within a P frame is mainly the motion estimation information which is essential for the decoding and display of the P and B frames. In the event that Key frame information is missing, i.e., a Key frame was skipped or lost, all the subsequent P frames based on that particular K frame will be ignored in order to prevent visual artifacts. The video server utilizes the fact that partial Key frame information is missing, based on feedback from the video client, to skip sending subsequent P frames that are based on the corrupted or lost Key frame in order to conserve bandwidth.

Detailed Description Text (26):
The raw video source is compressed into multiple types of frames comprised of video data having varying degrees of quality since the network cannot guarantee any

particular bandwidth or an error free network connection. Thus, these multiple frame types can be assigned varying degrees of importance or priority. The most important of all the frame types are the Key frames which are assigned the highest priority. Being the most important, key frames are sent using a reliable mechanism. Such a reliable mechanism includes using a network protocol such as TCP or reliable UDP. Reliable UDP refers to utilizing UDP, a basically unreliable protocol, in combination with a reliable mechanism that sits at a higher layer in the communication stack such as the Application Layer. The upper communication levels ensure that packets are delivered to the client.

Detailed Description Text (27):
The second most important frame type are the P frames which are transmitted using a semi reliable protocol such as reliable UDP as described above. If P frames are lost or corrupted en route to the video client, the video server may or may not resend them. For example, if too much time has passed, replacement packets would arrive at the client too late for display.

Detailed Description Text (28):
The least important frame type are the B frames which are sent using an unreliable protocol such as UJDP. The B frame data may or may reach the video client due to the condition of the network connection between the server and the client Upon arrival at the client of B frame data, the client determines whether it is useful and should be displayed. If the client determines that the B frame is not usable, an interpolgion mechanism is used to improve the video quality.

Detailed Description Text (29):
As described previously, the video steam stored in the video and audio source file (compressed video and audio file 16 in FIG. 1), is made up of three type of frames, i.e., Key framnes, P frames and B frames, that are grouped into sequences of GOPs. In addition, each frame type is fiter broken down into multiple levels of detail. In the example protocol and file format disclosed herein, each frame type is further broken down into five different video data levels, numbered 1 through 5. Level 1 contains the least amount of data which represents the lowest video quality and level 5 contains the most amount of data representing the highest quality of video.

Detailed Description Text (72):
The video compression/file generator 212 functions similarly to that of the video compression/file generator 14 of FIG. 1 with the exception that the video compression/file generator of FIG. 15 generates a separate compressed video/audio file for each compression level. For N compression levels, the video compression/file generator 212 functions to generate a compressed video/audio file 214 for levels 1 through N. Considering the system described previously, compressed video/audio files 214 are generated for Levels 1 through Level 5. The compressed video/audio files may be in any suitable format such as AVI format. The generation of the compressed video/audio files 214 can be performed either on-line or off-line. Typically the video/audio file is generated off-line. Note that any suitable method of video compression can be utilized to process the raw video data 210 such as described in connection with the MPEG-1, MPEG-2 or MPEG-4 standards.

Detailed Description Text (74):
Each of the N video servers 216 can comprise the video server 18 (FIG. 2) described previously or may comprise a standard off the shelf video server such as the MPEG-2 based Media Server from Oracle Inc. or the NetShow Server from Microsoft Corporation, Redmond, Wash. The standard video server must be suitably modified to provide a communication capability with the rate controller 222 before it will operate in the present invention. The modifications typically include providing a communication interface between the standard video server and the rate controller.

Detailed Description Paragraph Table (1):

Term Definition
AVI Audio Video Interlcaved CPU Central Processing Unit GOP Group of Picturcs GUI Graphical User Interface IP Internet Protocol ISDN Integrated Services Digital Network LAN Local Area Network MPEG Motion Picture Expert Group POTS Plain Old Telephone Service RSVP Reservation Protocol TCP Transmission Control Protocol UDP User Datagram Protocol

CLAIMS:

2. The method according to claim 1, wherein said step of compressing comprises the step of compressing the raw video source into a plurality of different types of frames, each frame type containing different amount of video content information, said plurality of different types of frames grouped so as to form a video stream consisting of a plurality of group of pictures (GOP) sequences.

3. The method according to claim 1, wherein said step of compressing comprises the step of compressing the raw video source into Key, P and B type frames, said Key, P and B frames generated so as to form a video stream consisting of a plurality of group of pictures (GOP) sequences.

4. A method of transporting video from a video server to a video client over a network channel, comprising the steps of:

compressing data from a raw video source so as to generate a plurality of frames, each frame being of a particular frame type, each frame type containing a particular amount of video content information, each frame comprising a plurality of levels, each level corresponding to a particular degree of compression;

estimating the bandwidth of the network channel;

determining the amount of video information waiting to be displayed at the video client;

selecting one of said plurality of levels of each frame to send over the network channel in accordance with said bandwidth estimate whereby the level selected optimizes the use of the bandwidth of the network channel;

choosing which frames having a particular frame type to send over the network channel in accordance with the amount of video information waiting to be displayed at the video client; and

sending the chosen frames having a particular frame type and of said selected level over the network channel.

5. A video server for transporting video from a video source over a network channel to a video client, said video source consisting of a plurality of frames of video data, each frame of video data consisting of multiple compression levels and being of a particular type, said video server comprising:

receiver means for inputting frames of video data from the video source;

sending means coupled to said receiver means, said sending means for determining which compression level within said frame and which frames having a particular type to transmit in accordance with the estimated available bandwidth of the network channel, said sending means for encapsulating said frames of video data into a plurality of packets for transmission over said network channel; and

a controller for managing the operation of said receiver means and said sending means whereby the rate of transmission of said sending means is maintained so as to

match the available bandwidth of the network channel.

6. The video server according to claim 5, wherein said sending means comprises:

a rate control unit for measuring the available bandwidth of the network channel;

a frame selector for inputting video frame data output by said receiver means, said frame selector outputting frames of a particular compression level in accordance with the bandwidth measured by said rate control unit;

a packet generator for inputting video frame data output by said frame selector, said packet generator for encapsulating said video frame data into a plurality of packets for transmission, said packet generator determining which frames having a particular type are to be transmitted;

a packet transmitter for placing onto the network channel the plurality of packets output by said packet generator; and

a receiver for receiving acknowledgments sent by the video client over the network channel in response to packets received thereby.

7. A method of transporting video from a video server to a video client over a network channel, comprising the steps of:

compressing data from a raw video source so as to generate a plurality of frames, each frame being of a particular frame type, each frame type containing a particular amount of video content information, each frame comprising a plurality of levels, each level corresponding to a particular degree of compression;

estimating the bandwidth of the network channel;

determining the amount of video information waiting to be displayed at the video client;

selecting one of said plurality of levels of each frame to send over the network channel in accordance with said bandwidth estimate whereby the level selected optimies the use of the bandwidth of the network channel;

choosing which frames having a particular frame type to send over the network channel in accordance with the amount of video information waiting to be displayed at the video client;

sending the chosen frames of a type containing a higher amount of video data content and of a selected level over the network channel utilizing a reliable communication protocol; and

sending the chosen frames of a type containing a lower amount video data content and of a selected level over the network channel utilizing an unreliable communication protocol.

8. A video server system for transporting video from a plurality of video sources over a network channel to a video client, each video source consisting of a plurality of frames of video data, each frame of video data consisting of a single compression level and being of a particular type, said video server system comprising:

a plurality of video servers, each video server associated with a single video source at a particular compression level, each video server comprising:

receiver means for inputting frames of video data from the video source associated

with that particular video server;

sending means coupled to said receiver means, said sending means for determining which frames having a particular type to transmit in accordance with the available bandwidth of the network channel, said sending means for encapsulating said frames of video data into a plurality of packets for transmission over said network channel;

a controller for managing the operation of said receiver means and said sending means; and

a rate controller for determining which video server to utilize for transmission of video data based on the available bandwidth of the network channel.

9. The video server system according to claim 8, wherein said sending means comprises:

means for interfacing said video server to said rate controller;

a bandwidth measurement unit for measuring the available bandwidth of the network channel;

a packet generator for inputting video frame data output by said receiver means, said packet generator for encapsulating said video frame data into a plurality of packets for transmission, said packet generator determining which frames having a particular type are to be transmitted;

a packet transmitter for placing onto the network channel the plurality of packets output by said packet generator; and

a receiver for receiving acknowledgments sent by the video client over the network channel in response to packets received thereby.

11. The method of claim 10, wherein step a) comprises compressing the raw video source into a plurality of different types of frames, each frame type having video data corresponding to one of the plurality of quality levels.

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L11: Entry 3 of 41 File: USPT Jul 1, 2003

DOCUMENT-IDENTIFIER: US 6587985 B1
TITLE: Data transmission method, data transmission apparatus, data receiving apparatus, and packet data structure

Abstract Text (1):
A data transmission apparatus including a receiving unit for receiving transmitted packets; a priority decision unit; a retransmission packet storage unit; a retransmission instruction receiving unit for receiving a retransmission request from a terminal at the receiving end; a retransmission decision unit; a transmission queue management unit; and a transmission unit.

Application Filing Date (1):
19991130

Brief Summary Text (4):
For transmission of video (audio and video) data on the Internet, a download type transmission method and a stream type transmission method are currently employed.

Brief Summary Text (18):
According to a first aspect of the present invention, there is provided a data transmission method for performing continuous data transmission from the transmitting end to the receiving end in units of packets, each packet having additional information relating to its sequence number, priority, and data reproduction time at the receiving end, while successively reproducing data of packets received at the receiving end. This method comprising the steps of: at the transmitting end, giving priority information to each packet to be transmitted; and storing, as retransmission data, only data of packets the priorities of which are equal to or higher than a predetermined value, in a retransmission buffer; at the receiving end, when a transmission error is detected, detecting the priority information of an error packet; and when the detected priority is equal to or higher than the predetermined value, outputting a retransmission request for the error packet to the transmitting end by indicating the sequence number of this error packet; at the transmitting end, only when the data of the packet having the sequence number which is indicated by the retransmission request from the receiving end is stored in the retransmission buffer, retransmitting the data of this packet to the receiving end; and discarding the data stored in the retransmission buffer in order starting from a packet which cannot be in time for data reproduction at the receiving end. Therefore, the transmission quality in a radio section in real-time transmission is improved and, further, the number of retransmission times is reduced.

Brief Summary Text (20):
According to a third aspect of the present invention, in the data transmission method of the first aspect, when the data transmitted from the transmitting end to the receiving end is video data based on MPEG, a packet which contains data corresponding to frames coded by utilizing intra-frame correlation is regarded as a packet having a high priority. Therefore, the transmission quality in a radio section in real-time transmission is improved and, further, the number of retransmission times is reduced.

Brief Summary Text (21):
According to a fourth aspect of the present invention, in the data transmission method of the first aspect, at the transmitting end, the additional information relating to the sequence number and the priority of a predetermined packet is also embedded in a subsequent packet to be transmitted after the predetermined packet; and at the receiving end, in the case where a transmission error has occurred in the predetermined packet and the additional information of the predetermined packet has an error, a retransmission request for the predetermined packet as an error packet is made on the basis of the additional information of the predetermined packet which is embedded in the subsequent packet, when the subsequent packet transmitted after the predetermined packet is received. Therefore, the transmission quality in a radio section in real-time transmission is improved and, further, the number of retransmission times is reduced.

Brief Summary Text (22):
According to a fifth aspect of the present invention, in the data transmission method of the fourth aspect, at the transmitting end, the process of embedding the sequence number of a predetermined high priority packet in a subsequent packet which follows the predetermined high priority packet is continuously performed until a high priority packet next to the predetermined high priority packet is transmitted; and at the receiving end, the sequence number of another packet which is embedded in the received packet is extracted, and when a transmission error has occurred in the packet of the extracted sequence number, a retransmission request for this error packet is made by indicating the sequence number of this packet. Therefore, the transmission quality in a radio section in real-time transmission is improved and, further, the number of retransmission times is reduced .

Brief Summary Text (23):
According to a sixth aspect of the present invention, there is provided a data transmission apparatus for relaying data which are successively transmitted from the transmitting end in units of packets, each packet having additional information relating to its sequence number, priority, data reproduction time at the receiving end. This apparatus comprises: a receiving unit for receiving the packets transmitted from the transmitting end; a priority decision unit for deciding the priority of each of the received packets; a retransmission packet storage unit for storing packets the priorities of which are equal to or higher than a predetermined value, as retransmission packets, on the basis of the priority of each packet decided by the priority decision unit; a retransmission instruction receiving unit for receiving a retransmission request from a terminal at the receiving end; a retransmission decision unit for deciding whether retransmission of the packet for which the retransmission request has been made should be performed or not, on the basis of the retransmission request and the storage status of the retransmission packets in the retransmission packet storage unit; a transmission queue management unit for setting the transmission order of the received packets and the packets which have been decided as packets to be retransmitted, on the basis of the additional information; and a transmission unit for transmitting the data of these packets in the transmission order set by the management unit. Therefore, only the error packets the priorities of which are equal to or higher than a predetermined value can be retransmitted, whereby the transmission quality of a radio section in real-time transmission is improved and, further, the number of retransmission times is reduced.

Brief Summary Text (24):
According to a seventh aspect of the present invention, there is provided a data receiving apparatus for receiving data which are transmitted from the transmitting end in units of packets, each packet having additional information relating to its sequence number, priority, and data reproduction time at the receiving end, and successively reproducing the data for each packet. This apparatus comprises: a receiving unit for receiving the packets transmitted from the transmitting end; an error packet detection unit for detecting error packets in which errors have

occurred during transmission, and outputting normal packets which have been transmitted without transmission errors, on the basis of the data of the received packets; a packet priority decision unit for receiving the output from the error packet detection unit, and deciding error packets the priorities of which are equal to or higher than a predetermined value; and a retransmission instruction output unit for outputting a retransmission request for each of the error packets the priorities of which are decided as being equal to or higher than the predetermined value, to the transmitting end, by indicating the sequence number of this error packet. Therefore, at the receiving end, a retransmission request to the transmitting end is made only for the error packet the priority of which is equal to or higher than a predetermined value, whereby the transmission quality in a radio section in real-time transmission is improved and, further, the number of retransmission times is reduced.

Brief Summary Text (25):
According to an eighth aspect of the present invention, there is provided a data transmission method in which data transmission from the transmitting end to the receiving end is continuously performed in units of packets, each packet having additional information relating to its sequence number, priority, and data reproduction time at the receiving end, while successively reproducing data of packets which have arrived at the receiving end and, at this time, only packets which can be in time for data reproduction at the receiving end are retransmitted. This method comprises: at the transmitting end, giving a data reproduction time at the receiving end to each packet to be transmitted; and storing, as retransmission data, only data of packets the priorities of which are equal to or higher than a predetermined value, in a retransmission buffer; at the receiving end, when a transmission error is detected, detecting the reproduction time for an error packet and the arrival time of the error packet, and deciding an arrival time limit in accordance with the reproduction time; and when the error packet has arrived before the arrival time limit, outputting a retransmission request for the error packet to the transmitting end by indicating the sequence number of this error packet; at the transmitting end, when the data of the packet having the sequence number indicated by the retransmission request from the receiving end is stored in the retransmission buffer, retransmitting data of the packet the transmission time of which does not pass the reproduction time, to the receiving end, while discarding data of the packet the transmission time of which has passed the reproduction time; and discarding the data stored in the retransmission buffer in order starting from a packet which cannot be in time for data reproduction at the receiving end. Therefore, the transmission quality in a radio section in real-time transmission is improved and, further, the number of retransmission times is reduced.

Brief Summary Text (29):
According to a twelfth aspect of the present invention, there is provided a data transmission apparatus for relaying data which are successively transmitted from the transmitting end in units of packets, each packet having additional information relating to its sequence number, priority, data reproduction time at the receiving end. This apparatus comprises: a receiving unit for receiving the packets transmitted from the transmitting end; a priority decision unit for deciding the priority of each of the received packets; a reproduction time decision unit for deciding packets which cannot be in time for reproduction at the receiving end, amongst the packets to be transmitted to the receiving end; a retransmission packet storage unit for storing packets the priorities of which are equal to or higher than a predetermined value, as retransmission packets, on the basis of the priority of each packet decided by the priority decision unit; a retransmission instruction receiving unit for receiving a retransmission request from a terminal at the receiving end; a retransmission decision unit for deciding whether retransmission of the packet for which the retransmission request has been made should be performed or not, on the basis of the retransmission request and the storage status of the retransmission packets in the retransmission packet storage unit; a transmission queue management unit for setting the transmission order of the

received packets and the packets which have been decided as packets to be retransmitted, on the basis of the additional information; and a transmission unit for transmitting, in the transmission order set by the management unit, the data of packets other than the packets which are decided as packets that cannot be in time for reproduction at the receiving end, by the reproduction time decision unit. Therefore, the transmission. quality in a radio section in real-time transmission is improved and, further, the number of retransmission times is reduced.

Brief Summary Text (30):
According to a thirteenth aspect of the present invention, there is provided a data receiving apparatus for receiving data which are transmitted from the transmitting end in units of pickets, each packet having additional information relating to its sequence number, priority, and data reproduction time at the receiving end, and successively reproducing the data for each packet. This apparatus comprises: a receiving unit for receiving the packets transmitted from the transmitting end; an error packet detection unit for detecting error packets in which errors have occurred during transmission, and outputting normal packets which have been transmitted without transmission errors, on the basis of the data of the received packets; a reproduction time decision unit for detecting the reproduction time given to each error packet detected by the error packet detection unit and the arrival time of the error packet at the receiving end, and setting the arrival time limit based on the reproduction time, and deciding whether or not the error packet has arrived at the receiving end before the arrival time limit; and a retransmission instruction output unit for outputting a retransmission request only for the error packet which has arrived at the receiving end before the, arrival time limit, to the transmitting end, by indicating the sequence number of the error packet, on the basis of the result of the decision in the reproduction time decision unit. Therefore, the transmission quality in a radio section in real-time transmission is improved and, further, the number of retransmission times is reduced.

Brief Summary Text (31):
According to a fourteenth aspect of the present invention, there is provided a data transmission method for performing continuous data transmission from the transmitting end to the receiving end in units of packets, each packet having additional information relating to its sequence number, priority, and data reproduction time at the receiving end, while successively reproducing data of packets arrived at the receiving end. This method comprises: at the transmitting end, giving a data reproduction time and priority information to each packet to be transmitted; and storing, as retransmission data, only data of packets the priorities of which are equal to or higher than a predetermined value, in a retransmission buffer; at the receiving end, when a transmission error is detected, detecting the priority information of an error packet, the reproduction time of the error packet, and the arrival time of the error packet; setting the arrival time limit of the error packet on the basis of the reproduction time; and when the detected priority is equal to or higher than the predetermined value and the error packet has arrived before the arrival time limit, outputting a retransmission request for this error packet to the transmitting end by indicating the sequence number of this error packet; at the transmitting end, when data of the packet having the sequence number indicated by the retransmission request from the receiving end is stored in the retransmission buffer, retransmitting only data of the packet the transmission time of which does not pass the reproduction time, to the receiving end, while discarding data of the packet the transmission time of which has passed the reproduction time; and discarding the data stored in the retransmission buffer in order starting from a packet which cannot be in time for reproduction at the receiving end. Therefore, the transmission quality in a radio section in real-time transmission is improved and, further, the number of retransmission times is reduced.

Brief Summary Text (32):

According to a fifteenth aspect of the present invention, in the data transmission method of the fourteenth aspect, at the transmitting end, additional information relating to the sequence number, the priority, and the reproduction time of a predetermined packet is embedded in a subsequent packet to be transmitted after the predetermined packet; and at the receiving end, when a transmission error of the predetermined packet has occurred and the additional information of the predetermined packet has an error, a retransmission request for the predetermined packet as an error packet is made on the basis of the additional information of the predetermined packet which is embedded in the subsequent packet, when the subsequent packet transmitted after the predetermined packet is received. Therefore, the transmission quality in a radio section in real-time transmission is improved and, further, the number of retransmission times is reduced.

Brief Summary Text (33):
According to a sixteenth aspect of the present invention, in the data transmission method of the fifteenth aspect, at the transmitting end, the process of embedding the sequence number of a predetermined high priority packet in a subsequent packet which follows the predetermined high priority packet is continuously performed until a high priority packet next to the predetermined high priority packet is transmitted; and at the receiving end, the sequence number of another packet which is embedded in the received packet is extracted, and when a transmission error has occurred in the packet of the extracted sequence number, a retransmission request for this packet is made by indicating the sequence number of this packet. Therefore, the transmission quality in a radio section in real-time transmission is improved and, further, the number of retransmission times is reduced.

Brief Summary Text (37):
According to a twentieth aspect of the present invention, there is provided a data transmission apparatus for relaying data which are successively transmitted from a distribution server, in units of packets each having additional information relating to its sequence number, priority, data reproduction time at the receiving end. This apparatus comprises: a receiving unit for receiving the packets transmitted from the transmitting end; a priority decision unit for deciding the priority of each of the received packets; a retransmission packet storage unit for storing packets the priorities of which are equal to or higher than a predetermined value, as retransmission packets, on the basis of the priority of each packet decided by the priority decision unit; a retransmission instruction receiving unit for receiving a retransmission request from a terminal at the receiving end; a retransmission decision unit for deciding whether retransmission of the packet for which the retransmission request has been made is to be performed or not, on the basis of the retransmission request and the storage status of the retransmission packets in the retransmission packet storage unit; a retransmission instruction output unit for outputting the retransmission request for the error packet requested by the terminal, to the distribution server, on the basis of the result of the decision in the retransmission decision unit; a transmission queue management unit for setting the transmission order of the received packets and the packets which have been decided as packets to be retransmitted, on the basis of the additional information; and a transmission unit for transmitting the data of these packets in the transmission order set by the management unit. Therefore, the number of retransmission times between the distribution server and the relay server can be reduced.

Brief Summary Text (38):
According to a twenty-first aspect of the present invention, there is provided a data transmission method for performing continuous data transmission from the transmitting end to the receiving end in units of packets, each packet having additional information relating to its sequence number, priority, and data reproduction time at the receiving end, while successively reproducing data of packets received at the receiving end. This method comprises: at the transmitting end, when a packet the priority of which is equal to or higher than a predetermined

value is transmitted as a high priority packet, storing data of this high priority packet, as retransmission data, in a retransmission buffer; managing the value of the transmitting end high priority sequence number which corresponds to the number of transmitted high priority packets, and the value of the sequence number of the high priority packet so that these values are correlated with each other; and transmitting a subsequent packet which follows the high priority packet after embedding the value of the transmitting end high priority sequence number in this subsequent packet; at the receiving end, extracting the value of the transmitting end high priority sequence number which is embedded in the received packet; managing the value of the receiving end high priority sequence number which corresponds to the number of received high priority packets; when the value of the extracted transmitting end high priority sequence number is not equal to the value of the receiving end high priority sequence number, outputting a retransmission request to the transmitting end, by indicating the value of the transmitting end high priority sequence number which is obtained on the basis of the value of the receiving end high priority sequence number; and updating the value of the receiving end high priority sequence number; at the transmitting end, only when data of the packet having the sequence number corresponding to the value of the transmitting end high priority sequence number which is indicated by the retransmission request from the receiving end is stored in the retransmission buffer, retransmitting the data of this packet to the receiving end. Therefore, retransmission of the high priority packet the priority of which is equal to or higher than a predetermined value, can be performed by simpler procedures.

Brief Summary Text (39):
According to a twenty-second aspect of the present invention, in the data transmission method of the twenty-first aspect, at the receiving end, when the value of the transmitting end high priority sequence number embedded in the received packet is not equal to the value of the receiving end high priority sequence number, a retransmission request is output to the transmitting end, by listing the values ranging from the value obtained by adding 1 to the receiving end high priority sequence number, to the value of the transmitting end high priority sequence number, as the values of the transmitting end high priority sequence numbers, or by designating the range as the range of the values of the transmitting end high priority sequence numbers; and at the transmitting end, the sequence numbers corresponding to the values of the plural transmitting end high priority sequence numbers which are indicated by the retransmission request from the receiving end are retrieved, and only when data of the packets having the sequence numbers obtained by the retrieval are stored in the retransmission buffer, the data of the packets are retransmitted to the receiving end. Therefore, retransmission of the high priority packet the priority of which is equal to or higher than a predetermined value, can be performed by simpler procedures.

Brief Summary Text (40):
According to a twenty-third aspect of the present invention, in the data transmission method of the twenty-first aspect, at the receiving end, the retransmission request is performed continuously several times, indicating the value of a transmitting end high priority sequence number; and at the transmitting end, the sequence number corresponding to the value of the transmitting end high priority sequence number which is indicated by the retransmission request from the receiving end is retrieved, and data of the packet having the sequence number obtained by the retrieval is retransmitted to the receiving end and, simultaneously, the correspondence between the value of the sequence number obtained by the retrieval and the value of the transmitting end high priority sequence number indicated by the receiving end is deleted. Therefore, when at least one of several transmission requests from the receiving end is normally received at the transmitting end, only the error packet the priority of which is equal to or higher than a predetermined value can be retransmitted, whereby the transmission quality in a radio section in real-time transmission can be effectively improved.

Brief Summary Text (41):
According to a twenty-fourth aspect of the present invention, there is provided a data transmission apparatus for relaying data which are successively transmitted from the transmitting end in units of packets, each packet having additional information relating to its sequence number, priority, data reproduction time at the receiving end. This apparatus comprises: a receiving unit for receiving the packets transmitted from the transmitting end; a transmission queue management unit for setting the transmission order of the received packets and packets which are decided as packets to be retransmitted; a transmission unit for transmitting data of these packets in the transmission order set by the transmission queue management unit; a priority decision unit for deciding the priority of each of the received packets; a retransmission packet storage unit for storing packets the priorities of which are equal to or higher than a predetermined value, as retransmission packets, on the basis of the priority of each packet decided by the priority decision unit; a sequence number management unit for managing the value of the transmitting end high priority sequence number which corresponds to the number of transmitted high priority packets, and the value of the sequence number of the high priority packet so that these values are correlated with each other; a high priority sequence number insertion unit for embedding the value of the transmitting end high priority sequence number in a subsequent packet which follows the high priority packet; a retransmission instruction receiving unit for receiving a retransmission request indicating the high priority sequence number, from a terminal at the transmitting end; and a retransmission decision unit for deciding whether retransmission of the packet for which the retransmission request has been made is to be performed or not, on the basis of the retransmission request and the storage status of the retransmission packets in the retransmission packet storage unit. Therefore, only the error packet the priority of which is equal to or higher than a predetermined value can be retransmitted, whereby retransmission of the high priority packet can be performed by simpler procedures.

Brief Summary Text (42):
According to a twenty-fifth aspect of the present invention, there is provided a data receiving apparatus for receiving data which are transmitted from the transmitting end in units of packets, each packet having additional information relating to its sequence number, priority, and data reproduction time at the receiving end, and successively reproducing the data for each packet. This apparatus comprises: a receiving unit for receiving the packets transmitted from the transmitting end; an error packet detection unit for detecting an error packet in which an error has occurred during transmission, and outputting a normal packet which has been transmitted without transmission errors, and the value of the transmitting end high priority sequence number which corresponds to the number of transmitted high priority packets and is embedded in the normal packet; a high priority sequence number management unit for managing the value of the receiving end high priority sequence number which corresponds to the number of normal high priority packets which have been received without transmission errors, on the basis of the output from the error packet detection unit; a retransmission sequence number decision unit for comparing the value of the transmitting end high priority sequence number from the error packet detection unit with the value of the receiving end high priority sequence number, and when these values are not equal, deciding the value of the transmitting end high priority sequence number for which a retransmission request is to be made, on the basis of the value of the receiving end high priority sequence number; and a retransmission instruction output unit for outputting a retransmission request to the transmitting end, by indicating the value of the decided transmitting end high priority sequence number. Therefore, retransmission of the high priority packet can be performed with simpler procedures.

Brief Summary Text (43):
According to a twenty-sixth aspect of the present invention, there is provided a data structure of a packet for performing data transmission from the transmitting

end and the receiving end, wherein the packet comprises a header section containing relevant information indicating the attribute of the packet, and a data section containing data to be transmitted; and the header section comprises at least first and second header information, amongst first header information indicating the sequence number corresponding to the packet, second header information indicating the priority of the packet, and third header information indicating the reproduction time at the receiving end, of the data to be transmitted. Therefore, retransmission of a low priority packet and retransmission of a packet which cannot be in time for reproduction can be avoided, whereby the transmission quality in a radio section in real-time transmission is improved while reducing the number of retransmission times.

Brief Summary Text (45):
According to a twenty-eighth aspect of the present invention, in the packet data structure of the twenty-seventh aspect, the header section of the packet includes the first and second information or the first and third information, as attribute information of a packet which has already been transmitted before the packet. Therefore, retransmission control based on the priority or the reproduction time can be performed with reliability.

Brief Summary Text (46):
According to a twenty-ninth aspect of the present invention, in the packet data structure of the twenty-sixth aspect, the header section of the packet includes the value of the high priority sequence number corresponding to the number of high priority packets which have been transmitted before the packet and having the priorities equal to or higher than a predetermined value. Therefore, the transmission quality in a radio section in real-time transmission is improved and, moreover, retransmission of an error packet is realized by simpler procedures.

Detailed Description Text (9):
By the way, in the retransmission control required for the real-time video data transmission, even when a transmission error occurs while predetermined packets are transmitted and thereby several frames of images are lost, this is not a fatal transmission error which leads to abnormal end of data transmission. Accordingly, in the video data transmission method, to complete real-time transmission with a measure of reliability is given the highest priority.

Detailed Description Text (10):
For example, in transmission of a video signal based on MPEG standard, in the case where a packet corresponding to an I frame (intra-frame coded image) as a reference image becomes an error packet, even when subsequent packets corresponding to a P frame (inter-frame forward-prediction coded image) and a B frame (inter-frame bidirectional-prediction coded image) are received normally, video signals of the P frame and the B frame cannot be reproduced. So, as for the I frame, it is necessary to recover the transmission error to the utmost.

Detailed Description Text (13):
First retransmission control is selective retransmission control for reducing the number of retransmission times by selecting, as packets to be retransmitted, high priority packets amongst the error packets. Second retransmission control is retransmission control with time limit for reducing excessive retransmission by stopping retransmission of packets which cannot be in time for reproduction, amongst the error packets.

Detailed Description Text (17):
In the data transmission method of this first embodiment, data transmission from the transmitting end to the receiving end is continuously performed in units of packets, each packet having additional information relating to its sequence number, priority, and data reproduction time at the receiving end, while successively reproducing data of packets received at the receiving end. At this time, only error

packets the priorities of which are equal to or higher than a predetermined value are retransmitted.

Detailed Description Text (20):
Each packet transmitted from the transmitting end is composed of a data section containing digital data such as video data, audio data, and text data, and a header section containing additional information other than these digital data. To be specific, the header section of each packet contains additional information relating to its sequence number, priority, and data reproduction time at the receiving end.

Detailed Description Text (21):
Further, the data transmission apparatus 101 includes a buffer 17 for retransmission (hereinafter, referred to as a retransmission buffer 17), a packet priority decision unit 15, and a retransmission buffer management unit 18. The retransmission buffer 17 stores predetermined packets amongst the received packets, as retransmission packets. The packet priority decision unit 15 decides the priorities of the received packets. The retransmission buffer management unit 18 controls the retransmission buffer 17 such that data of packets the priorities of which are equal to or higher than a predetermined value are stored in the buffer 17, in accordance with the decided priorities of the packets. To be specific, only when data of a packet having a sequence number indicated by a retransmission request from the receiving side is stored in the retransmission buffer 17, the retransmission decision unit 16 decides that data of this packet should be retransmitted to the receiving end.

Detailed Description Text (23):
While in FIG. 1(a) the data transmission apparatus 101 constitutes a relay server, when the data transmission apparatus is a distribution server, it is constructed as shown in FIG. 1(b). That is, in FIG. 1(b), the receiving unit 11 of the data transmission apparatus 101 is replaced with a coded packet generation unit 10a which encodes the data and outputs the coded data in packet units, and a priority allocation unit 10b which allocates additional information such as a priority to each packet output from the coded packet generation unit 10a.

Detailed Description Text (26):
Further, the data receiving apparatus 201 includes a reception history management unit 24, a packet priority decision unit 25, and a retransmission instruction output unit 26. The reception history management unit 24 manages the packet reception history. The packet priority decision unit 25 receives the result of the detection in the error packet detection unit 22 and decides an error packet the priority of which is equal to or higher than a predetermined value. The retransmission instruction output unit 26 outputs a request for retransmitting the error packet which has been decided in the packet priority decision unit 25, toward the transmitting end, by indicating the sequence number of the error packet.

Detailed Description Text (29):
In the data transmission method of this first embodiment, when a transmission error occurs during packet transmission, a retransmission request is made for only the packets the priorities of which are equal to or higher than a predetermined value, from the receiving end to the transmitting end, while no retransmission request is made for the error packets the priorities of which are lower than the predetermined value.

Detailed Description Text (30):
For example, assuming that the priorities equal to or higher than the predetermined value are high priorities while the priorities lower than the predetermined value are low priorities, when an error occurs during transmission of a high priority packet (S1) of sequence number S1, a retransmission request for this high priority packet (S1) is made. However, when an error occurs during transmission of a low

priority packet (S2) of sequence number S2, no retransmission request is made for this low priority packet (S2).

Detailed Description Text (31):
To be specific, each packet transmitted from the distribution server is given additional information relating to its sequence number and priority. In the data transmission apparatus 101 as a relay server, the transmission order of the received packets is set by the transmission queue management unit 12, and the packets are supplied to the transmission unit 13. On the other hand, the priorities of the received packets are decided by the packet priority decision unit 15. Then, in the transmission unit 13, transmission of these packets is performed according to the transmission order which has been set. Further, those packets the priorities of which are decided as being equal to or higher than the predetermined value are stored in the retransmission buffer 17 under control of the retransmission buffer management unit 17. Further, in the retransmission buffer 17, data are successively released (discarded) from the packets which cannot be in time for reproduction, under control of the management unit 18.

Detailed Description Text (34):
In the data receiving apparatus 201, the packets from the relay server (data transmission apparatus) 101 are received by the receiving unit 21, and the received packets are supplied to the error packet detection unit 22. Then, only the packets which have been transmitted without transmission errors are output from the error packet detection unit 22 to the packet decoding unit 23, and the additional information of each packet is supplied to the reception history management.unit 24. At this time, the priority information of each error packet is supplied to the packet priority decision unit 25, wherein it is decided whether or not the priority of the error packet is equal to or higher than a predetermined value. With respect to the error packet the priority of which is equal to or higher than the predetermined value, the retransmission instruction output unit 26 outputs a retransmission request to the transmitting end, by indicating the sequence number of this error packet.

Detailed Description Text (36):
As described above, according to the first embodiment of the present invention, data transmission from the transmitting end to the receiving end is continuously performed in units of packets each having additional information relating to its sequence number, priority, and data reproduction time and, simultaneously, data of the packets received at the receiving end are successively reproduced. With respect to error packets affected by transmission errors, only those having priorities equal to or higher than a predetermined value are retransmitted. Therefore., the transmission quality of the radio section in the real-time transmission is improved and, moreover, the number of retransmission times can be reduced.

Detailed Description Text (37):
In this first embodiment, each packet may be given the frame type, such as I frame, P frame, and B frame, as the additional information, instead of the priority.

Detailed Description Text (38):
Further, there are various methods for deciding the packet priority. For example, in the case of a video signal based on the MPEG standard, packets corresponding to I frames may be decided as high priority packets.

Detailed Description Text (39):
Further, in the packet discarding process performed when the retransmission buffer is filled to the capacity, the above-described first or second updating process may be performed on the packets in the order of ascending priorities.

Detailed Description Text (43):
The data transmission apparatus 102 includes an error correction unit 31, in

addition to the constituents of the data transmission apparatus 101 of the first embodiment. The error correction unit 31 performs an error correction process in which each packet output from the transmission queue management unit 12 is given error correction codes for additional information such as its sequence number, priority, etc., and the packet which has been subjected to the error correction process is supplied to the transmission unit 13. Other constituents of the data transmission apparatus 102 are identical to those of the data transmission apparatus 101 of the first embodiment.

Detailed Description Text (47):
In the data transmission method according to the second embodiment, at the transmitting end, each packet to be transmitted is given error correction codes for the additional information relating to its sequence number, priority, etc. At the receiving end, the additional information is subjected to error correction according to the error correction codes and, thereafter, a retransmission request for the error packet is made in accordance with the additional information. Thereby, even when the sequence number and the priority information have errors, a retransmission request for the error packet can be correctly performed.

Detailed Description Text (50):
In the data transmission method of the third embodiment, at the transmitting end, additional information relating to the sequence number and priority of a predetermined packet is embedded in a subsequent packet which will be transmitted after the predetermined packet. At the receiving end, when a transmission error of the predetermined packet occurs and thereby the additional information of the predetermined packet has an error, a retransmission request for the error packet is made when receiving the subsequent packet which is transmitted after the error packet, in accordance with the additional information of the predetermined packet which is embedded in the subsequent packet.

Detailed Description Text (51):
For example, assuming that the priorities equal to or higher than a predetermined value are high priorities while the priorities lower than the predetermined value are low priorities, as shown in FIG. 6, when an error occurs during transmission of Ma high priority packet (S1) of sequence number S1 and only a low priority packet (S2) of sequence number S2 which follows the packet S1 is normally received, a retransmission request for the high priority packet (S1) is made when the next low priority packet (S2) is received.

Detailed Description Text (52):
On the other hand, when an error occurs during transmission of a low priority packet (S3) of sequence number S3 and only a high priority packet (S4) of sequence number S4 which follows the packet S3 is normally received, no retransmission request for the low priority packet S3 is made when the next high priority packet (S4) is received.

Detailed Description Text (53):
In the data transmission method so constructed, even when an error occurs during transmission of a predetermined packet and thereby the sequence number or the priority information of this packet has an error, since the additional information (sequence number, priority, etc.) of the predetermined packet is embedded in the subsequent packet which is transmitted next to this packet, a transmission request for this error packet (predetermined packet) can be made correctly.

Detailed Description Text (57):
The data transmission apparatus 103 includes a sequence number storage unit 33 and a sequence number insertion unit 32, in addition to the constituents of the data transmission apparatus 101 of the first embodiment. The sequence number storage unit 33 stores the sequence numbers of the packets the priorities of which are equal to or higher than a predetermined value, amongst the packets transmitted by a

transmission unit 13a. The sequence number insertion unit 32 outputs each of the sequence numbers stored in the sequence number storage unit 33 to the transmission unit 13a so that the sequence number is inserted in the header of the packet to be transmitted. Further, the transmission unit 13a of this modification is different from the transmission unit 13 of the first embodiment only in that it inserts the sequence number supplied from the sequence number insertion unit 32 into the header of the packet supplied from the transmission queue management unit 12, before transmitting the packet. Other constituents of the data transmission apparatus 103 are identical to those of the data transmission apparatus 101 of the first embodiment.

Detailed Description Text (59):
The data receiving apparatus 203 according to the modification of the third embodiment includes an inserted sequence extraction unit 42, in addition to the constituents of the data receiving apparatus 201 of the first embodiment. The inserted sequence extraction unit 42 extracts, from a normal packet output from the error packet detection unit 22, the sequence number of a high priority packet which has been received in advance of the normal packet. The normal packet is output to the packet decoding unit 23 through the inserted sequence extraction unit 42. Further, in the data receiving apparatus 203, the packet priority decision unit 25a outputs a retransmission request to the retransmission instruction output unit 26 when the packet of the sequence number extracted by the inserted sequence extraction unit 42 is an error packet. Other constituents of the data receiving unit 203 are identical to those of the data receiving apparatus 201 of the first embodiment.

Detailed Description Text (62):
In the data transmission apparatus (transmitting end) 103, in addition to the transmission operation of the data transmission apparatus 101 according to the first embodiment, the process of embedding the sequence number of a high priority packet to be transmitted in the subsequent packets, is carried out until the next high priority is transmitted.

Detailed Description Text (63):
For example, assuming that the priorities equal to or higher than a predetermined value are high priorities while the priorities lower than the predetermined value are low priorities, as shown in FIG. 9, after a high priority packet (S1) of sequence number S1 has been transmitted, subsequent packets (S2).about.(S4) of sequence numbers S2.about.S4 are transmitted after the sequence number S1 of the previous high priority packet (S1) is embedded therein, and then a high priority packet (S5) of sequence number S5 is transmitted after the sequence number S4 of the previous high priority packet (S4) is embedded therein.

Detailed Description Text (64):
When an error occurs while the high priority packet (S1) and the subsequent low priority packets (S2) and (S3) are transmitted and so only the high priority packet (S4) is normally received, a retransmission request for the first high priority packet S1 is made when the next high priority packet (S4) is received.

Detailed Description Text (66):
In the data transmission method according to the modification of the third embodiment, at the transmitting end, the process of embedding the sequence number of a high priority packet, the priority of which is equal to or higher than a predetermined value, into the subsequent packets which follow this high priority packet, is continued until a high priority packet next to the high priority packet is transmitted. At the receiving end, the sequence number of another packet (high priority packet) embedded in the received packet is extracted. When this packet (another packet) is an error packet, a retransmission request for this packet is made by indicating the sequence number of this packet. Therefore, even when two successive packets become error packets, the sequence number of the high priority

packet which has become an error packet can be detected from the header information of the subsequent packet which is transmitted without a transmission error, whereby a retransmission request for the high priority error packet can be made with higher reliability.

Detailed Description Text (67):
In the third embodiment, the sequence number and priority information of each packet are embedded in the header of a packet to be transmitted next to this packet, and in the modification of the third embodiment, the sequence number of a high priority packet to be transmitted is embedded in the subsequent plural packets until the next high priority packet is transmitted. However, the information to be embedded in the subsequent packet is not restricted thereto. For example, the number of retransmission times may be embedded in the subsequent packet together with the sequence number.

Detailed Description Text (73):
The data transmission: apparatus 104 includes a sequence number/retransmission count storage unit 35, a sequence number/retransmission count insertion unit 34, and a retransmission count clear unit 38. The sequence number/retransmission count storage unit 35 stores the sequence numbers and the retransmission counts, of the packets the priorities of which are equal to or higher than a predetermined value, amongst the packets transmitted by the transmission unit 13a. The sequence number/retransmission count insertion unit 34 outputs the sequence number and the retransmission count which are stored in the storage unit 35, to the transmission unit 13a, such that these data are inserted in the header of the packet to be transmitted. The retransmission count clear unit 38 subjects the packet received at the receiving unit 11 to a process of clearing the retransmission count, and outputs the packet to the transmission queue management unit 12. Further, the transmission unit 13a of this fourth embodiment is different in function from the transmission unit 13 of the first embodiment only in that it inserts the sequence number and the retransmission count supplied from the sequence number/retransmission count insertion unit 34 into the header of the packet supplied from the transmission queue management unit 12 and then transmits this packet.

Detailed Description Text (83):
In the state shown in FIG. 12, the high priority packet of sequence number S1 has already been received normally at the receiving end. Hereinafter, a packet having a sequence number Sn and retransmission count N is represented as a packet (Sn,N) [n,N: natural numbers]. In the figure, packets of sequence numbers S1, S2, and S4 are high priority packets, and packets of sequence numbers S3 and S5 are packets other than the high priority packets. In this fourth embodiment, only the additional information of the above-mentioned high priority packets are embedded in the subsequent packets.

Detailed Description Text (84):
Initially, as shown in FIG. 12, at a predetermined transmission timing, the high priority packet (S2,1) subsequent to the high priority packet (S1) is retransmitted. At this time, in the header of the packet (S2,1), the sequence number S1 and retransmission count 1 of the high priority packet (S1) are embedded as well as its sequence number S2 and retransmission count 1.

Detailed Description Text (85):
At the next transmission timing, the packet (S3) subsequent to the high priority packet (S2,1) is transmitted, having, in its header, the sequence number S2 and retransmission count 1 of the high priority packet (S2,1) as well as its sequence number S3.

Detailed Description Text (86):
At the next transmission timing, the next high priority packet (S4,1) is

retransmitted, having, in its header, the sequence number S2 and retransmission count 1 of the high priority packet (S2,1) as well as its sequence number S4 and retransmission count 1.

Detailed Description Text (87):
At the next transmission timing, the next packet (S5) is transmitted, having, in its header, the sequence number S4 and retransmission count 1 of the high priority packet (S4,1) as well as its sequence number S5.

Detailed Description Text (89):
In this state, at the transmitting end, it is known that the high priority packet which has been received most recently is the packet having the sequence number S1, and the error packet is the packet having the sequence number S4 and the retransmission count 1, but it is not known what kinds of packets have been transmitted between the high priority packet (S1) and the packet (S5).

Detailed Description Text (90):
So, the receiving end sends a retransmission request for the high priority packet (S4,1), together with the sequence number S1 of the most-recently received high priority packet as well as the sequence number S4 and retransmission count 1 of this packet (S4,1).

Detailed Description Text (91):
Then, at the transmitting end, the sequence number S1 of the most-recently received high priority packet is compared with the sequence number S4 of the error packet for which the retransmission request has been made. In this case, since the sequence number of the most-recently received high priority packet is smaller than the sequence number of the error packet, the transmitting end performs selective retransmission for those packets having sequence numbers larger than the sequence number S1 and equal to or smaller than the sequence number S4.

Detailed Description Text (92):
In this case, the high priority packet (S2,2) is transmitted, having, in its header, the sequence number S4 and retransmission count 1 of the high priority packet (S4,1) and, subsequently, the high priority packet (S4,2) is transmitted, having, in its header, the sequence number S2 and retransmission count 2 of the high priority packet (S2,2).

Detailed Description Text (93):
On receipt of the high priority packet (S2,2), the receiving end sends a retransmission request for the high priority packet (S4,1) toward the transmitting end, together with the sequence number S2 of the most-recently received high priority packet (S2,2) as well as the sequence number S4 and retransmission count 1 of this packet (S4,1). However, with respect to the high priority packet (S4), since the second retransmission has already been done, no retransmission is performed in response to the retransmission request for the high priority packet (S4,1).

Detailed Description Text (95):
In the case shown in FIG. 13, data exchange from transmission of the high priority packet (S2,1) to transmission of the high priority packet (S2,2) is identical to that described with respect to FIG. 12.

Detailed Description Text (96):
In the case shown in FIG. 13, transmission of the high priority packet (S2,2) is error transmission, and the next high priority packet (S4,2) is normally received.

Detailed Description Text (97):
In this case, at the receiving end, it is known that the high priority packet which has been received most recently has the sequence number S4, and the error packet

has the sequence number S2 and the retransmission count 2.

Detailed Description Text (98):
So, the receiving end sends a retransmission request for the high priority packet (S2,2), together with the sequence number S4 of the most-recently received high priority packet as well as the sequence number S2 and retransmission count 2 of this packet (52,2).

Detailed Description Text (99):
Then, at the transmitting end, the sequence number S4 of the most-recently received high priority packet is compared with the sequence number S2 of the error packet for which the retransmission request has been made. In this case, since the sequence number S4 of the most-recently received high priority packet is larger than the sequence number S2 of the error packet, the transmitting end performs selective retransmission for only the error packet.

Detailed Description Text (100):
That is, the high priority packet (S2,3) is transmitted, having, in its header, the sequence number S4 and retransmission count 2 of the high priority packet (S4,2).

Detailed Description Text (102):
Initially, as shown in FIG. 14, at a predetermined transmission timing, the high priority packet (S4,1) is retransmitted, having, in its header, the sequence number S2 and retransmission count 1 of the high priority packet (S2,1) which has been transmitted in advance of this packet, as well as the sequence number S4 and retransmission count 1 of this packet.

Detailed Description Text (103):
At the next transmission timing, the next packet (S5) is transmitted, having, in its header, the sequence number S4 and retransmission count 1 of the high priority packet (S4,1) as well as the sequence number S5 of this packet.

Detailed Description Text (105):
In this state, at the receiving end, it is known that the most-recently received high priority packet has the sequence number S1, and the error packet has the sequence number S4 and the retransmission count 1

Detailed Description Text (106):
So, the receiving end sends a retransmission request for the high priority packet (S4,1), together with the sequence number S1 of the most-recently received high priority packet as well as the sequence number S4 and retransmission count 1 of this packet.

Detailed Description Text (107):
Then, at the transmitting end, the sequence number S1 of the most-recently received high priority packet is compared with the sequence number S4 of the error packet for which the retransmission request has been made. In this case, since the sequence number S1 of the most-recently received high priority packet is smaller than the sequence number S4 of the error packet, the transmitting end performs selective retransmission for those packets having sequence numbers larger than the sequence number S1 and equal to or smaller than the sequence number S4.

Detailed Description Text (108):
In this case, the high priority packet (S2,2) is transmitted, having, in its header, the sequence number S4 and retransmission count I of the high priority packet (S4,1) as well as the sequence number S2 and retransmission count 2 of this packet (S2,2) and, subsequently, the high priority packet (S4,2) is transmitted, having, in its header, the sequence number S2 and retransmission count 2 of the high priority packet (S2,2) as well as the sequence number S4 and transmission count 2 of this packet.

Detailed Description Text (109):
Thereafter, the packet of sequence number 56 is transmitted, having, in its packet, the sequence number S4 and retransmission count 2 of the high priority packet (S4,2) as well as the sequence number S6 of this packet.

Detailed Description Text (110):
Since errors have occurred during transmission of the high priority packets (S2,2) and (S4,2), these packets are not received at the receiving end, and only the packet (S6) is received.

Detailed Description Text (111):
In this state, at the receiving end, it is known that the most-recently received high priority packet has the sequence number S1, and the error packet has the sequence number S4 and the retransmission count 2.

Detailed Description Text (112):
So, the receiving end sends a retransmission request for the high priority packet (S4,2), together with the sequence number S1 of the most-recently received high priority packet as well as the sequence number S4 and retransmission count 2 of this packet.

Detailed Description Text (113):
Then, at the transmitting end, the sequence number S1 of the most-recently received high priority packet is compared with the Subsequence number S4 of the error packet for which the retransmission request has been made. In this case, since the sequence number S1 of the most-recently received high priority packet is smaller than the sequence number S4 of the error packet, the transmitting end performs selective retransmission for those packets having sequence numbers which are larger than the sequence number S1 and equal to or smaller than the sequence number S4.

Detailed Description Text (114):
That is, the high priority packet (S2,3) is transmitted, having, in its header, the sequence number S4 and retransmission count 2 of the high priority packet (S4,2) as well as the sequence number S2 and retransmission count 3 of this packet and, subsequently, the high priority packet (S4,3) is transmitted, having, in its header, the sequence number S2 and retransmission count 3 of the high priority packet (S2,3) as well as the sequence number S4 and retransmission count 3 of this packet.

Detailed Description Text (115):
As described above, according to the fourth embodiment, when the sequence number of the most-recently received high priority packet is larger than the sequence number of the error packet, the transmitting end retransmits only the error packet. Therefore, retransmission of the error packet can be performed efficiently.

Detailed Description Text (119):
Initially, as shown in FIG. 15, at a predetermined transmission timing, the high priority packet (S1,1) is transmitted, having, in its header, the sequence number and retransmission count of a previous high priority packet as well as the sequence number S1 and retransmission count 1 of this packet. In this case, there are two receiving ends for the transmitting end, i.e., the receiving end 1 and the receiving end 2, and transmission of the high priority packet (S1,1) is error transmission.

Detailed Description Text (120):
In this case, each receiving end sends a retransmission request for the high priority packet (S1,1), together with the sequence number (lasts) of the most-recently received high priority packet.

Detailed Description Text (121):
In response to the retransmission request from the receiving end 1, the transmitting end retransmits the high priority packet (S1,2). However, after this retransmission, the transmitting end does not perform retransmission in response to the retransmission request from the receiving end 2, because the high priority packet (S1,2) has already been retransmitted in response to the retransmission request from the receiving end 2.

Detailed Description Text (122):
As the result of the retransmission of the high priority packet (S1,2), the receiving end 1 receives the high priority packet (S1) while the receiving end 2 does not receive the high priority packet (S1).

Detailed Description Text (123):
In this case, the receiving end 2 outputs a retransmission request for the high priority packet (S1,2). In response to this request, the transmitting end increments the retransmission count and retransmits the high priority packet (S1,3).

Detailed Description Text (124):
In the fourth embodiment and the modification thereof, an upper limit may be set for the retransmission count. Further, the upper limit of the retransmission count may be changed according to the priority value of the packet.

Detailed Description Text (127):
According to the data transmission method of this fifth embodiment, in the data transmission method of the first embodiment, absence of a sequence number is detected at the receiving end, and a retransmission request for a packet having the absent sequence number is made by using this sequence number. At the transmitting end, when the packet of the sequence number for which the retransmission request has been output from the receiving end is stored in the retransmission buffer 17, retransmission of this packet is performed. At the transmitting end, only high priority packets are stored in the retransmission buffer 17.

Detailed Description Text (133):
When a high priority packet (S1) of sequence number S1, which has been output from the data transmission apparatus (transmitting end) 105, is not received by the data receiving apparatus (receiving end) 205 due to a transmission error, in the data receiving apparatus 205, the error packet detection unit 22 detects that the sequence number S1 is absent, and a retransmission request for the packet of this sequence number S1 is output to the transmitting end.

Detailed Description Text (134):
At the transmitting end, it is decided whether the packet of the retransmission request is stored in the retransmission buffer 17 or not. In this case, since the packet of the retransmission request is a high priority packet, it is stored in the retransmission buffer 17. Therefore, the transmitting end performs retransmission of this packet.

Detailed Description Text (136):
At the transmitting end, it is decided whether the packet of the retransmission request is stored in the retransmission buffer 17 or not. In this case, since the packet of the retransmission request is not a high priority packet, it is not stored in the retransmission buffer 17. Therefore, the transmitting end does not perform retransmission of this packet.

Detailed Description Text (140):
In the data transmission method of this sixth embodiment, data transmission from the transmitting end to the receiving end is continuously performed in units of packets each having additional information relating to its sequence number,

priority, and data reproduction time and, simultaneously, data in the packets received at the receiving end are successively reproduced. At this time, only the packet which can arrive at the receiving end within the time limit is retransmitted.

Detailed Description Text (145):
The data receiving apparatus 206 includes a reproduction time decision unit 43, instead of the packet priority decision unit 25 of the data receiving apparatus 201 of the first embodiment. The reproduction time decision unit 43 detects the reproduction time which is given to the error packet detected by the error packet detection unit 22 and the arrival time of the error packet at the receiving end, decides the arrival time limit based on the reproduction time, and decides whether or not the error packet has arrived at the receiving end before the arrival time limit. On the basis of the result of the decision, the retransmission instruction output unit 26 instructs the transmitting end to retransmit the error packet which has arrived at the receiving end before the arrival time limit, by using the sequence number of the packet. The decision of the arrival time limit by the reproduction time decision unit 43 is performed based on at least one of the allowable packet delay time which is decided at the receiving end, and the packet transmission delay time between the transmitting end and the receiving end.

Detailed Description Text (151):
For example, when high priority packets (S1) and (S2) having sequence numbers S1 and S2, which have been transmitted from the data transmission apparatus (transmitting end) 106, are not normally received at the data receiving apparatus (receiving end) 206 due to a transmission error, in the data receiving apparatus 206, the error packet detection unit 22 decides that these high priority packets (S1) and (S2) are error packets. Further, the reproduction time decision unit 43 detects the reproduction times (T1) and (T2) and the arrival times of these error packets, decides the arrival time limits (T1+.alpha.) and (T2+.alpha.) in accordance with the reproduction times, and decides whether or not these error packets have arrived at the receiving end before the arrival time limits, respectively.

Detailed Description Text (152):
Since the high priority packet (S1) has arrived at the receiving end before the time limit, the receiving end instructs the transmitting end to retransmit this packet.

Detailed Description Text (153):
On the other hand, since the high priority packet (S2) has not arrived at the receiving end before the time limit, the receiving end does not instruct the transmitting end to retransmit this packet.

Detailed Description Text (155):
As described above, according to the sixth embodiment of the present invention, data transmission from the transmitting end to the receiving end is continuously performed in units of packets each having additional information relating to its sequence number, priority, and data reproduction time at the receiving end and, simultaneously, data of the packets received at the receiving end are successively reproduced. At this time, only the packets which have arrived at the receiving end within the time limit at the receiving end, are retransmitted to the transmitting end. Therefore, the transmission quality of wireless section in real-time transmission is improved and, moreover, the number of retransmission times can be reduced.

Detailed Description Text (170):
For example, the second embodiment in which each packet is given error correction codes for its sequence number and priority information may be combined with the first modification of the sixth embodiment in which each packet is given error

correction codes for its sequence number and reproduction time (first combination). According to this combination, each packet is given error correction codes for its sequence number, priority information, and reproduction time.

Detailed Description Text (171):
Further, the third embodiment in which the sequence number and the priority information of a predetermined packet are embedded in a packet to be transmitted next to the predetermined packet, may be combined with the second modification of the sixth embodiment in which the sequence number and the reproduction time of a predetermined packet are embedded in a packet to be transmitted next to the predetermined packet (second combination). According to this combination, the sequence number, the priority information, and the reproduction time of the predetermined packet are embedded in the packet to be transmitted next to the predetermined packet.

Detailed Description Text (172):
Further, the modification of the third embodiment in which the process of embedding the sequence number and the priority information of a high priority packet to be transmitted into subsequent packets is continued until a next high priority packet is transmitted, may be combined with the second modification of the sixth embodiment in which the sequence number and the reproduction time of a predetermined packet are embedded in a packet to be transmitted next to the predetermined packet (third combination). According to this combination, the process of embedding the sequence number, the priority information, and the reproduction time of a high priority packet into subsequent packets is continued until a next high priority packet is transmitted.

Detailed Description Text (173):
Furthermore, according to the combination of the selective retransmission control and the retransmission control with time limit (second or third combination), the quantity of data to be embedded in the packet increases. Therefore, there is proposed a method of embedding a difference between the header information (sequence number, priority information, reproduction time) of a predetermined packet and the header information of a subsequent packet, in the subsequent packet.

Detailed Description Text (181):
In the third modification of the sixth embodiment so constructed, since a difference between the header information (sequence number, priority information, reproduction time) of a predetermined packet and the header information of a subsequence packet is embedded in the subsequence packet, the quantity of information to be embedded in the packet is reduced.

Detailed Description Text (182):
While in the third modification of the sixth embodiment the sequence number and the reproduction time are described as information to be embedded in the subsequent packet, information to be embedded is not restricted thereto. For example, in addition to the sequence number and the reproduction time, the retransmission count and the priority may be embedded.

Detailed Description Text (211):
Furthermore, like the data transmission apparatus according to embodiment 1 or 6, the data transmission apparatus 108 has the function of storing a packet the priority of which is equal to or higher than a predetermined value in a retransmission buffer and discarding a packet which cannot be in the time for retransmission, and the function of retransmitting a packet for which a retransmission request is output from the receiving end, although these functions are not shown in the figure.

Detailed Description Text (212):

To be specific, the data transmission apparatus 108 includes the constituents corresponding to the retransmission instruction receiving unit 14, the packet priority decision unit 15, the retransmission decision unit 16, the retransmission buffer 17, and the retransmission buffer management unit 18 which are included in the data transmission apparatus 101 of the first embodiment. In the transmission queue management unit 12, setting of the packet transmission order is performed on all the packets to be transmitted, including not only the received packets but also the packets to be transmitted.

Detailed Description Text (217):
To be specific, the data receiving apparatus 208 includes the constituents corresponding to the error packet detection unit 22, the reception history management unit 24, the packet priority decision unit 25, and the retransmission instruction output unit 26 which are included in the data receiving apparatus 201 of the first embodiment.

Detailed Description Text (233):
FIGS. 29-31 are diagrams for explaining a data transmission method according to a ninth embodiment of the present invention. In the data transmission method of this ninth embodiment, data transmission from the transmitting end to the receiving end is continuously performed in units of packets each having additional information relating to its sequence number, priority, and data reproduction time at the receiving end, which are required to realize real-time transmission in packet units, while successively reproducing data of the packets received at the receiving end. At this time, only error packets the priorities of which are equal to or higher than a predetermined value are retransmitted.

Detailed Description Text (236):
The data transmission apparatus 109 further includes a high priority sequence number management unit 81, a sequence number correspondence management unit 82, and a high priority sequence number insertion unit 83. When the priority of a packet transmitted by the transmission unit 13 is equal to or higher than a predetermined value, the high priority sequence number management unit 81 increments the value of the sequence number which corresponds to only the high priority packet (high priority sequence number), and stores the incremented value. The sequence number correspondence management unit 82 stores the correspondence between the value of the sequence number of the transmitted high priority packet and the value of the incremented high priority sequence number of this high priority packet. The high priority sequence number insertion unit 83 outputs the value of the high priority sequence number of each high priority packet, which is stored in the management unit 81, so that it is inserted in the packet to be transmitted.

Detailed Description Text (237):
The high priority sequence numbers managed by the high priority sequence number management unit 81 correspond to the number of high priority packets transmitted from the transmitting end.

Detailed Description Text (238):
Further, the data transmission apparatus 109 includes a retransmission buffer 17, a packet priority decision unit 15, and a retransmission buffer management unit 18. The retransmission buffer 17 stores predetermined packets amongst the received packets, as retransmission packets. The packet priority decision unit 15 decides the priorities of the received packets. The retransmission buffer management unit 18 controls the retransmission buffer 17 such that data of packets the priorities of which are equal to or higher than a predetermined value are stored in the buffer 17, in accordance with the decided priorities of the packets.

Detailed Description Text (239):
Further, the data transmission apparatus 109 includes a retransmission instruction receiving unit 14 and a retransmission decision unit 16c. The retransmission

instruction receiving unit 14 receives a retransmission request indicating a high priority sequence number, from the terminal at the receiving end. The retransmission decision unit;16c decides whether retransmission of a packet for which the retransmission request has been made is performed or not. The retransmission decision unit 16 retrieves the management information in the sequence number correspondence management unit 82, in accordance with the high priority sequence number indicated by the retransmission request, to obtain the sequence number corresponding to the high priority sequence number used for the retransmission request, and decides that the requested packet is to be retransmitted, only when the packet of the sequence number is stored in the retransmission buffer 17.

Detailed Description Text (240):
In FIG. 29(a), real-time data transmission is performed in packet units between the distribution server and the terminal through the relay server or the like, and the data transmission apparatus 109 constitutes the relay server. However, the relay server may serve as the distribution server. To be specific, when the data transmission apparatus serves as the distribution server, it is constructed as shown in FIG. 29(b). In FIG. 29(b), the receiving unit 11 of the data transmission apparatus 109 is replaced with a coded packet generation unit 10a which encodes the data and outputs the coded data in packet units, and a priority allocation unit 10b which allocates additional information, such as a priority, to each packet output from the coded packet generation unit 10a.

Detailed Description Text (243):
When the high priority sequence number inserted in the packet from the data transmission apparatus (transmitting end high priority sequence number) is correctly extracted, the error packet detection unit 22a outputs both of the value of the transmitting end high priority sequence number and the value of the receiving end high priority sequence number at this point of time. The value of the receiving end high priority number corresponds to the number of the high priority packets received flat the receiving end, and this value is incremented every time a high priority packet is received at the receiving end.

Detailed Description Text (244):
Further, the data receiving apparatus 209 includes a high priority sequence number management unit 91 and a retransmission sequence number decision unit 92. When the error packet detection unit 22a outputs a normal packet, the high priority sequence number management unit 91 increments the value of the receiving end high priority sequence number and stores it. The retransmission sequence number decision unit 92 compares the value of the transmitting end high priority sequence number output from the error packet detection unit 22a with the value of the receiving end high priority sequence number. When these values are not equal, the decision unit 92 outputs the values ranging from the value obtained by adding 1 to the value of the receiving end high priority sequence number to the value of the transmitting end high priority sequence number, as the values of retransmission sequence numbers (transmitting end high priority sequence numbers).

Detailed Description Text (245):
The high priority sequence number management unit 91 increments the value of the stored receiving end high priority sequence number every time the retransmission sequence number decision unit 92 outputs a high priority sequence number.

Detailed Description Text (246):
Further, the data receiving unit 209 includes a retransmission instruction output unit 26c which outputs a retransmission request for an error packet to the transmitting end, on the basis of the transmitting end high priority sequence number which is output as a retransmission sequence number from the retransmission sequence number decision unit 92.

Detailed Description Text (249):
In the description with respect to FIG. 31, a sequence number [S+n] indicates a sequence number having a value "S+n", a sequence number [H+n] indicates a high priority sequence number having a value "H+n", and a packet (S+n) indicates a packet having a sequence number the value of which is "S+n". Further, n is any of 0, 1, 2, 3, 4, and 5.

Detailed Description Text (250):
For example, assuming that the priorities equal to or higher than the predetermined value are high priorities while the priorities lower than the predetermined value are low priorities, as shown in FIG. 31, when an error has occurred during transmission of high priority packets (S+1) and (S+2) of sequence numbers [S+1] and [S+2], retransmission requests for these high priority packets are made at the receiving end. However, when an error has occurred during transmission of a low priority packet (S+3) of sequence number [S+3], no retransmission request is made for this low priority packet (S+3).

Detailed Description Text (251):
To be specific, each packet transmitted from the distribution server is given additional information relating to its sequence number and priority. In the data transmission apparatus 109 as a relay server, the transmission order of the received packets is set by the transmission queue management unit 12, and then the packets are supplied to the transmission unit 13. On the other hand, the priorities of the received packets are decided by the packet priority decision unit 15. Then, in the transmission unit 13, transmission of these packets is performed according to the transmission order which has been set. Further, those packets the priorities of which are decided as being equal to or higher than the predetermined value are stored in the retransmission buffer 17 under control of the retransmission buffer management unit 18. Further, in the retransmission buffer 17, data are successively released (discarded) from the packets which cannot be in time for reproduction, under control of the management unit 18.

Detailed Description Text (253):
When transmitting a high priority packet, the high priority sequence number is incremented.

Detailed Description Text (254):
To be specific, when a high priority packet (S+0) of sequence number [S+0] is transmitted by the transmission unit 13, the value of the transmitting end high priority sequence number [H+0] which is stored in the high priority sequence number management unit 81 is incremented to "H+1". At this time, the value of the sequence number [S+0] of the high priority packet (S+0) and the incremented value of the transmitting end high priority sequence number [H+1] are entered, by one-to-one correspondence, in the sequence number correspondence management unit 82.

Detailed Description Text (255):
Likewise, when a high priority packet (S+1) of sequence number [S+1] is transmitted by the transmission unit 13, the value of the transmitting end high priority sequence number [H+1] which is stored in the high priority sequence number management unit 81 is incremented to "H+2". At this time, the value of the sequence number [S+1] of the high priority packet (S+1) and the incremented value of the transmitting end high priority sequence number [H+2] are entered, by one-to-one correspondence, in the sequence number correspondence management unit 82.

Detailed Description Text (256):
Further, also when a high priority packet (S+2) is transmitted, like the high priority packets (S+0) and (S+1), the transmitting end high priority sequence number [H+2] in the high priority sequence number management unit 81 is incremented, and the sequence number [S+2] of the high priority packet (S+2) and the incremented transmitting end high priority sequence number [H+3] are entered,

by one-to-one correspondence, in the sequence number correspondence management unit 82.

Detailed Description Text (257):
On the other hand, when transmitting a low priority packet, the corresponding low priority sequence number is not incremented.

Detailed Description Text (258):
To be specific, when a low priority packet (S+3) of sequence number [S+3] is transmitted by the transmission unit 13, the value of the transmitting end high priority sequence number [H+3] stored in the high priority sequence number management unit 81 is not updated but maintained as it is. At this time, the process of entering the sequence number of the transmitted packet and the transmitting end high priority sequence number [H+3] which is stored in the high priority sequence number management unit 81, in the sequence number correspondence management unit 82, is not performed.

Detailed Description Text (260):
When the high priority packet (S+1) is transmitted, the value of the transmitting end high priority sequence number [H+1] which is stored in the high priority sequence number management unit 81 at this point of time, is embedded in the header of the transmission packet (S+1) by the high priority sequence number insertion unit 83. Likewise, when the low priority packet (S+3) is transmitted, the value of the transmitting end high priority sequence number [H+3] which is stored in the high priority sequence number management unit 81 at this point of time, is embedded in the header of the transmission packet (S+3) by the high priority sequence number insertion unit 83. Thereafter, the transmission packet having the transmitting end high priority sequence number so embedded in its header, is transmitted to the receiving end by the transmission unit 13.

Detailed Description Text (262):
In the data receiving apparatus 209, the packets from the relay server (data transmission apparatus) 109 are received by the receiving unit 21, and the received packets are supplied to the error packet detection unit 22a. The normally received high priority packet (S+0) is output from the error packet detection unit 22a to the packet decoding unit 23, and the value of its receiving end high priority sequence number (i.e., the value of the high priority sequence number [H+0] stored in the high priority sequence number management unit 91) is incremented to "H+1".

Detailed Description Text (263):
It is assumed that a transmission error has occurred during transmission of the high priority packets (S+1) and (S+2) and, thereafter, the low priority packet (S+3) subsequent to these packets has been transmitted without a transmission error.

Detailed Description Text (264):
In this case, the normally transmitted low priority packet (S+3) is output from the error packet detection unit 22a to the packet decoding unit 23, but the value of the receiving end high priority sequence number [H+1] which is stored in the high priority sequence number management unit 91 is not incremented.

Detailed Description Text (265):
Further, in the error packet detection unit 22a, when the transmitting end high priority sequence numbers [H+0] and [H+3] which are inserted in the high priority packet (s+0) and the low priority packet (S+3), respectively, are correctly extracted, these transmitting end high priority sequence numbers [H+0) and (H+3] are output to the retransmission sequence number decision unit 92. Further, the receiving end high priority sequence numbers [H+0] and [H+1] which are stored in the high priority sequence number management unit 91 at the time when the transmitting end high priority sequence numbers [H+0] and [H+3] are extracted by

the error packet detection unit 22a, are output to the retransmission sequence number decision unit 92.

Detailed Description Text (266):
For example, at the time when the transmitting end high priority sequence number [H+0] is extracted, the value [H+0] of the receiving end high priority sequence number stored in the high priority sequence number management unit 91 as well as the transmitting end high priority sequence number [H+0] are output to the retransmission sequence number decision unit 92. At the time when the transmitting end high priority sequence number [H+3] is extracted, the value [H+1] of the receiving end high priority sequence number stored in the high priority sequence number management unit 91 as well as the transmitting end high priority sequence number [H+3] are output to the retransmission sequence number decision unit 92.

Detailed Description Text (267):
In the retransmission sequence number decision unit 92, the transmitting end high priority sequence number and the receiving end high priority sequence number, which have been supplied at the same time, are compared, to decide whether retransmission is to be requested to the transmitting end.

Detailed Description Text (268):
For example, as the result of the comparison between the transmitting end high priority sequence number [H+0] and the receiving end high priority sequence number [H+0], since the values of these high priority sequence numbers are equal, no transmission instruction is performed. On the other hand, as the result of the comparison between the transmitting end high priority sequence number [H+3] and the receiving end high priority sequence number [H+1], since the values of these high priority sequence numbers are not equal, a retransmission instruction is performed. In this case, the values ranging from the value obtained by adding 1 to the value of the receiving end high priority sequence number [H+1] to the value of the transmitting end high priority sequence number [H+3], i.e., "H+2" and "H+3", are output to the retransmission instruction output unit 26 as the values of the high priority sequence numbers used for the retransmission instruction. At this time, in the high priority sequence number management unit 91, the value of the stored receiving end high priority sequence number is incremented twice to be "H+3".

Detailed Description Text (269):
On receipt of "H+2" and "H+3" as the values of the high priority sequence numbers, the retransmission instruction output unit 26 outputs a retransmission request with the high priority sequence number [H+2] and a retransmission request with the high priority sequence number [H+3], to the transmitting end.

Detailed Description Text (270):
Then, in the data transmission apparatus 109 at the transmitting end, the retransmission requests are received by the retransmission instruction receiving unit 14, and the management information in the sequence number correspondence management unit 82 is retrieved on the basis of the requested high priority sequence numbers [H+2] and [H+3], thereby obtaining the sequence number [S+1] corresponding to the high priority sequence number [H+2] and the sequence number [S+2] corresponding to the high priority sequence number [H+3]

Detailed Description Text (271):
Further, in the retransmission decision unit 16c, it is decided whether the data of the packets corresponding to the sequence numbers [S+1] and [S+2] are stored in the retransmission buffer 17 or not. Based on the result of this decision, only the packets the data of which are stored in the retransmission buffer 17 are output as retransmission packets from the retransmission buffer 17 to the transmission queue management unit 12. Here, the high priority packets (S+4) and (S+2) are output as retransmission packets.

Detailed Description Text (272):
In the retransmission queue management unit 12, the transmission order is set for these retransmission packets, and these packets are retransmitted to the receiving end through the transmission unit 13. Since the retransmission packets (S+1) and (S+2) are high priority packets, when transmitting these packets, the values of their transmitting end high priority sequence numbers stored in the high priority sequence number management unit 81 are incremented.

Detailed Description Text (273):
To be specific, when transmitting the retransmission packet (S+1), the value of the transmitting end high priority sequence number [H+3] stored in the high priority sequence number management unit 81 is incremented to "H+4", and the sequence number [S+1] of the retransmission packet (S+1) and the transmitting end high priority sequence number [H+4] are entered, by one-to-one correspondence, in the sequence number correspondence management unit 82.

Detailed Description Text (274):
Further, when transmitting the retransmission packet (S+2), the value of the transmitting end high priority sequence number [H+4] stored in the high priority sequence number management unit 81 is incremented to "H+5", and the sequence number [S+2] of the retransmission packet (S+2) and the transmitting end high priority sequence number [H+5] are entered, by one-to-one correspondence, in the sequence number correspondence management unit 82.

Detailed Description Text (275):
As described above, according to the ninth embodiment of the invention, data transmission from the transmitting end to the receiving end is continuously performed in units of packets, each packet having additional information relating to its sequence number, priority, and data reproduction time, and information relating to the high priority sequence number managed at the transmitting end and, simultaneously, data of received packets are successively reproduced at the receiving end. The value of the transmitting end high priority sequence number possessed by the received packet (number of transmitted high priority packets) is compared with the value of the receiving end high priority sequence number managed at the receiving end (number of received high priority packets), and a retransmission request is made by indicating a transmitting end high priority sequence number which is absent. Therefore, the transmission quality of the ratio section in real-time transmission can be improved by retransmission of error packets the priorities of which are equal to or higher than a predetermined value and, moreover, the retransmission of error packets can be realized by simpler procedures.

Detailed Description Text (276):
In this ninth embodiment, when the high priority sequence numbers corresponding to plural high priority packets transmitted are continuously absent, the receiving end sends a retransmission request for each high priority packet having the absent high priority sequence number, to the transmitting end. However, retransmission requests for plural high priority packets may be sent collectively to the transmitting end, by listing the values of the plural high priority sequence numbers or indicating the range of these values.

Detailed Description Text (277):
In this case, at the transmitting end, based on the plural high priority sequence numbers requested from the receiving end, the sequence numbers corresponding to the respective transmitting end high priority packets are obtained by retrieval, and the high priority packets having the sequence numbers so obtained are retransmitted to the receiving end.

Detailed Description Text (281):
The data transmission apparatus 110 includes a retransmission decision unit 16d,

instead of the retransmission decision unit 16c of the data transmission apparatus 109 of the ninth embodiment. The retransmission decision unit 16d performs the same process as that of the decision unit 16c and, further, outputs the sequence number of the packet which is decided to be transmitted. Further, the data transmission apparatus 110 includes a sequence number correspondence management unit 82a, instead of the sequence number correspondence management unit 82 of the data transmission apparatus 109. The sequence number correspondence management unit 82a performs the same process as that of the unit 82 and, further, deletes the value of the transmitting end high priority sequence number corresponding to the value of the sequence number supplied from the retransmission decision unit 16d. Other constitutes of the data transmission apparatus 110 of this tenth embodiment are identical to those of the data transmission apparatus 109 of the ninth embodiment.

Detailed Description Text (290):
On the other hand, when the high priority sequence number output from the retransmission sequence output unit 92 is input to the retransmission instruction output unit 26b, a retransmission request indicating the high priority sequence number is output from the retransmission instruction output unit 26b to the transmitting end and, simultaneously, this retransmission request is output as a control signal to the retransmission instruction consecutive output unit 93.

Detailed Description Text (293):
Further, at the transmitting end, packet retransmission is carried out according to the retransmission request, and the sequence number correspondence management unit 82a deletes the value of the transmitting end high priority sequence number corresponding to the sequence number supplied from the retransmission decision unit 16d.

Detailed Description Text (294):
Therefore, with respect to the same retransmission request which is received again, retrieval for the sequence number corresponding to this retransmission request by the retransmission decision unit 16d ends in failure because the correspondence between the transmitting end high priority sequence number indicated by this retransmission request and the sequence number is deleted from the sequence number correspondence management unit 82a.

Detailed Description Text (295):
As the result, in the retransmission decision unit 16d, it is decided that no retransmission is performed for the high priority packet for which the retransmission request has been made by indicating the value of the transmitting end high priority sequence number. Therefore, at the transmitting end, it is avoided that the same packet is repeatedly retransmitted when the same retransmission request is output several times from the receiving end.

Detailed Description Text (296):
As described above, according to the tenth embodiment of the present invention, a retransmission request indicating the high priority sequence number of a desired packet is consecutively transmitted several times, from the receiving end to the transmitting end, against transmission errors. Therefore, when at least one of the several transmission requests from the receiving end is normally received at the transmitting end, the error packet the priority of which is equal to or higher than a predetermined value can be retransmitted, whereby the transmission quality in the radio section in real-time transmission can be effectively improved.

Detailed Description Text (300):
The header section Ph includes header information Ia indicating the sequence number corresponding to each packet, header information Ib indicating the reproduction time at the receiving end, of the data to be transmitted (time stamp) Ib, header information indicating the priority of each packet, extension header information Id, and other header information I1 to I10 (refer to FIG. 34(b)).

Detailed Description Text (301):
The specific convention of each header information is described in RFC1889 as shown in FIG. 34(c). For example, the header information I3 indicates that the extension header information Id is added to the header section Ph when its value X is 1. The header information I5 indicates that the data stored in the data section is coded data by the MPEG1 method when its value PT is 32, and indicates that the data stored in the data section is coded data by the MPEG2 method when PT is 33. Further, each of the header information I9, Ic, I10, and I11 is header information to be added when the MPEG1 coded data is transmitted by RTP. The value P (P=1) of the header information Ic indicates that the data in the data section is I frame data, and the packet containing this I frame data is to be treated as a high priority packet. The value P (P=2) indicates that the data in the data section is P frame data, and the packet containing this P frame data is to be treated as a low priority packet. The value P(P=3) indicates that the data in the data section is B frame data, and the packet containing this B frame data is to be treated as a low priority packet.

Detailed Description Text (302):
Further, the extension header information Id corresponds to the sequence number and the priority information of the previous packet in the third embodiment (refer to FIG. 6), the sequence number and the retransmission count of the previously transmitted high priority packet in the fourth embodiment (refer to FIGS. 13 and 14), the sequence number and the reproduction time of the previous packet in the second modification of the sixth embodiment (refer to FIG. 21), the difference value of the sequence number of the previous packet and the difference value of the reproduction time of the previous packet in the third modification of the sixth embodiment (refer to FIG. 22), and the transmitting end high priority sequence number in the ninth embodiment (refer to FIG. 29).

CLAIMS:

1. A data transmission method for performing continuous data transmission from a transmitting end to a receiving end in units of packets, each packet having additional information relating to its sequence number, priority, and data reproduction time at the receiving end, while performing retransmission of the packets in response to a retransmission request from the receiving end, said method comprising: at the transmitting end, giving priority information to each packet to be transmitted; and at the receiving end, when a transmission error is detected, detecting the priority information of an error packet; when the detected priority is equal to or higher than a predetermined value, outputting a retransmission request for the error packet to the transmitting end by indicating the sequence number of the error packet; and at the transmitting end, retransmitting the data of the packet to the receiving end, the packet having the sequence number which is indicated by the retransmission request from the receiving end.

3. The data transmission method of claim 1 wherein, when the data transmitted from the transmitting end to the receiving end is video data based on MPEG, a packet which contains data corresponding to frames coded by utilizing intra-frame correlation is regarded as a packet having a high priority.

4. The data transmission method of claim 1 wherein: at the transmitting end, error correction codes relating to the additional information is further given to each packet to be transmitted; and at the receiving end, when a transmission error is detected, the additional information is subjected to an error correction process by using the error correction codes, and the priority information of an error packet is detected.

5. The data transmission method of claim 1 wherein: at the transmitting end, the additional information relating to the sequence number and the priority of a

previously transmitted packet is also embedded in a subsequent packet to be transmitted after a previously transmitted packet; and at the receiving end, in the case where a transmission error has occurred in a predetermined packet, a retransmission request for the predetermined packet as an error packet is made based on the additional information of the predetermined packet which is embedded in the subsequent packet transmitted after the predetermined packet, when the subsequent packet is received.

6. The data transmission method of claim 5 wherein: at the transmitting end, the process of embedding the sequence number of a predetermined high priority packet in a subsequent packet which follows the predetermined high priority packet is continuously performed until a high priority packet next to the predetermined high priority packet is transmitted; and at the receiving end, the sequence number of another packet which is embedded in the received packet is extracted, and when a transmission error has occurred in the packet of the extracted sequence number, a retransmission request for the packet of the extracted sequence number is made by indicating the sequence number of this packet.

7. A data transmission method for performing continuous data transmission from a transmitting end to a receiving end in units of packets, each packet having additional information relating to its sequence number, priority, and data reproduction time at the receiving end, while performing retransmission of the packets in response to a retransmission request from the receiving end, said method comprising: at the transmitting end, giving priority information to each packet to be transmitted; and storing, as retransmission data, only data of packets the priorities of which are equal to or higher than a predetermined value, in a retransmission buffer; at the receiving end, when a transmission error is detected, outputting a retransmission request for an error packet to the transmitting end by indicating the sequence number of the error packet; and at the transmitting end, only when the data of the packet having the sequence number which is indicated by the retransmission request from the receiving end is stored in the retransmission buffer, retransmitting the data of the packet having the sequence number which is indicated by the retransmission request to the receiving end.

11. The data transmission method of claim 7 wherein, when the data transmitted from the transmitting end to the receiving end is video data based on MPEG, a packet which contains data corresponding to frames coded by utilizing intra-frame correlation is regarded as a packet having a high priority.

12. A data transmission apparatus for relaying data transmitted from a transmitting end in units of packets, each packet having additional information relating to its sequence number, priority, and data reproduction time at a receiving end, said data transmission apparatus comprising: a receiving unit operable to receive packets transmitted from the transmitting end, a priority decision unit operable to decide the priorities of the received packets, a retransmission packet storing unit operable to store, as retransmission packets, only packets the priorities of which are equal to or higher than a predetermined value, based on the priority of each packet decided by the priority decision unit, a retransmission instruction receiving unit operable to receive a retransmission instruction from the receiving end, a retransmission decision unit operable to decide whether retransmission of a packet for which the retransmission instruction has been made is performed or not, based on the retransmission instruction and a storage condition of the retransmission packets in the retransmission packet storing unit, a transmission queue management unit operable to set the transmission order of the received packets and the packets being decided to be retransmitted in accordance with the additional information, and a transmission unit operable to transmit data of the packets in the transmission order set by the transmission queue management unit.

15. A data receiving apparatus for receiving data transmitted from a transmitting end in units of packets, each packet having additional information relating to its

sequence number, priority, and data reproduction time at a receiving end, said data receiving apparatus comprising: a receiving unit operable to receive packets transmitted from the transmitting end, an error packet detection unit operable to detect error packets in which errors have occurred during transmission and output normal packets which have been normally transmitted, a packet priority decision unit operable to receive the result of the detection in the error packet detection unit and decide an error packet the priority of which is equal to or higher than a predetermined value, and a retransmission instruction output unit operable to output a request for retransmitting the error packets toward the transmitting end by indicating the sequence number of the error packet, the error packets being decided to be packets the priorities of which are equal to or higher than the predetermined value in the packet priority decision unit.

17. The data receiving apparatus of claim 15 further comprising: an additional information extraction unit operable to, when receiving data transmitted from the transmitting end, in which the additional information relating to the sequence number and the priority corresponding to a previously transmitted packet is also embedded in a subsequent packet transmitted after the previously transmitted packet, extract the additional information of the previously transmitted packet embedded in the subsequent packet and inform the packet priority detecting unit of the extracted additional information, wherein, when the additional information of a predetermined packet has an error, said packet priority decision unit decides the error packet based on the additional information which is embedded in the subsequent packet and informed by the additional information extraction unit.

WEST Search History

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DATE: Tuesday, February 24, 2004

Hide?	Set Name	Query	Hit Count
		DB=PGPB,USPT,USOC,EPAB,JPAB,DWPI,TDBD; PLUR=YES; OP=ADJ	
☐	L11	L10 NOT l8	41
☐	L10	L9 NOT l5	45
☐	L9	l7 and MPEG	47
☐	L8	l7 and shape and texture and motion	5
☐	L7	l4 and ((frame near8 type) or (data near8 content) or feedback) and (priority or prioritization)	143
☐	L6	l3 and ((frame near8 type) or (data near8 content) or feedback) and (priority or prioritization)	2
☐	L5	l3 and ((frame near8 type) or (data near8 content) or feedback)	3
☐	L4	19991215	972
☐	L3	19991215	48
☐	L2	((video adj2 (Internet or IP)) or VoIP).ti.	370
☐	L1	(video adj2 (Internet or IP)) or VoIP	5374

END OF SEARCH HISTORY